

# NAVAL POSTGRADUATE SCHOOL Monterey, California



## THESIS

### SPEECH COMPRESSION USING COSINE PACKET DECOMPOSITION

by

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March 1996

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As digitization of data becomes more prevalent, the demands on existing communications networks and computer systems to cope with this increase in data become overwhelming. Currently, the speech compression problem is handled using the CELP( Code Excited Linear Prediction) scheme and its derivatives. Such techniques are the most frequently used for speech compression at medium-to-low rate ranges. Recent research conducted into the area of cosine packets has proven this field to be readily adaptable to speech compression and coding. In this thesis, speech compression schemes are developed using cosine-packet decomposition, minimum entropy basis selection, and an adaptive thresholding scheme for selecting coefficients. In addition, voiced-unvoiced segmentation and a denoising scheme were implemented. Test results showed high compression ratios (1:50) with a good quality of reconstructed speech.

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**SPEECH COMPRESSION USING COSINE PACKET DECOMPOSITION**

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## ABSTRACT

As digitization of data becomes more prevalent, the demands on existing communications networks and computer systems to cope with this increase become overwhelming. Currently, the speech compression problem is handled using the CELP (Code Excited Linear Prediction) scheme and its derivatives. Such techniques are the most frequently used for speech compression at medium-to-low rate ranges. Recent research conducted into the area of cosine packets has proven this field to be readily adaptable to speech compression and coding. In this thesis, speech compression schemes are developed using cosine-packet decomposition, minimum entropy basis selection, and an adaptive thresholding scheme for selecting coefficients. In addition, voiced-unvoiced segmentation and a denoising scheme are implemented. Test results show high compression ratios (1:50) with a good quality of reconstructed speech.







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# I. INTRODUCTION

Speech compression allows smaller bandwidth, higher data rates or a combination of these attributes. It can also be used to store speech like data in a compact form.

This thesis develops speech compression schemes based on the Local Trigonometric Transform [2], which use an adaptive thresholding scheme proposed in this work. These schemes perform a time partition of the original speech data first, according to a maximum depth selected by the user. An experimentally derived, optimum depth is proposed, based on the results of tests with several words and phonemes (defined in Chapter II). Following the time partitioning, a basis obtained via the minimum entropy best basis algorithm is selected. In order to perform compression, coefficients are selected according to an adaptive thresholding scheme, which varies the compression percentage depending on the energy and frequency content of each interval. The intervals are classified by a voiced-unvoiced segmentation algorithm. Depending on their classification, selection of coefficients is made in such a way that more coefficients are preserved for the voiced than for the unvoiced intervals. Then, these coefficients are encoded using uniform quantizers and Huffman coding to achieve average compression ratios of 1:50. In addition, two denoising schemes are proposed to minimize effects of equipment noise below 120 Hz, thereby improving the sound quality.

In a typical scenario, users of the proposed schemes will be able to adjust speech quality and transmission bandwidth, based on the current channel bandwidth available. They will be provided with the parameters that maximize the compression ratio, and minimize the required bandwidth at an acceptable speech quality. Using lower bit rate coding reduces the transmission bandwidth of the signal and may prove to be quite useful in partial band jamming environments where the available channel bandwidth may be limited. It is understood that the schemes proposed may be useful for military applications where the understanding of the message is more important than the overall quality of the sound. This thesis concentrates on finding the best possible compression

ratio, while still keeping an acceptable sound quality. In this work, sound quality is defined in terms of a mean square error as well as in terms of a proposed quality measure. Extensions of the proposed techniques lead to data storage improvements and they can also easily be adopted to cryptographic applications. The thesis is organized in the following manner. Chapter II presents an introduction to speech processing, where the concepts of phonemes and coarticulation effects are introduced and illustrated. Chapter III introduces the Local Trigonometric Transform and presents the Local Cosine Transform adopted in our work. The Local Cosine Transform can be viewed as a basic building block for the more complex Cosine Packet Transform, which has been used recently in speech applications [2]. The Cosine Packet Transform can also be viewed as a dual operation of the Wavelet Packet Transform [2]. Both packet schemes are presented, discussed and compared in Chapter IV. The Wavelet and Cosine Packet Transforms involve the selection of a particular basis “best” matched to the signal under study for compression applications. This choice of basis is carried out via the Best Basis algorithm, which is presented in Chapter V. Chapter VI presents the denoising and compression schemes investigated in this work. Denoising allows for enhancement of the audio quality of the speech signals when noise is present. Chapter VII describes the encoding schemes used to compress the speech information. Chapter VIII first discusses the experiments and parameters used to test our denoising and compression schemes. Next, it presents the results obtained using various phonemes, words and sentences. The data base consists of a limited collection of American-English words, some Portuguese words and some typical voiced and unvoiced sound segments. Some of the more elaborate data sets consist of complete sentences and dialogues. Finally, we compare compression results obtained with our Cosine Packet scheme and those obtained with the Wavelet Packet scheme using a “Daubechies” basis function [17]. Results show that the Cosine Packet Transform outperforms the Wavelet Packet Transform on the speech segments considered in this study. Finally, Chapter IX contains the conclusions and final considerations. All computer algorithms are listed in the Appendix.

## II. INTRODUCTION TO SPEECH PROCESSING

One of the principal differentiating features of any speech sound is excitation [1]. Two elemental excitation types are present in speech data: (1) voiced and (2) unvoiced. Voiced sounds have high energy and low frequency, while unvoiced sounds have low energy and high frequency. Another important characteristic of speech signals is that they are locally stationary.

The basic theoretical unit for describing how speech conveys linguistic meaning is called a *phoneme*. Each language has its own set of phonemes. For example, American English has about 42 phonemes, while Brazilian Portuguese has about 51 phonemes (Rio de Janeiro region). They are made of vowels, semivowels, diphthongs, and consonants. In general, the duration of each phoneme may vary from 15 to 400 milliseconds, depending on the sound produced and the way it is pronounced. For example, vowels can vary largely in duration, typically from 40 to 400 milliseconds.

The transition from one phoneme to another is not made abruptly or independently of adjacent phonemes. Actually, adjacent phonemes have a strong influence on the manner in which the transition takes place. The term used to refer to the change in phoneme articulation and acoustics that is caused by the influence of another phoneme is *coarticulation*.

Since this research investigates speech compression, there are two main requirements. First, we need to be able to split a speech signal into its smallest locally stationary “cells” constituted by phonemes, and represent them in a minimal way with good fidelity. Second, we need to preserve coarticulation effects as much as possible (i.e., we need to preserve the smooth transition from one phoneme to the next) .

Figure 2.1 illustrates the coarticulation process for the sound /issos/. The top plot represents time-domain speech. The middle plot represents the voiced and unvoiced portions of this sound obtained using the zero-crossing rate and the short-time energy contained in the sound [1]. The unvoiced portions are -ss- and -s-, corresponding to the

phoneme /s/. The high frequency and low energy of unvoiced segments are illustrated by the low short-time energy and high zero-crossing rates. The voiced portions of the sound are the phonemes /i/ and /o/. The low frequency and high energy of voiced phonemes are illustrated by the high short-time energy and low zero-crossing rate. The bottom plot shows the spectrogram obtained using a Hanning time window of length 256 samples with an overlap of 128. Note the coarticulation effects present, which allow for smooth transitions between phonemes. For example, the transition from /i/ to /s/ occurs through a “link,” which takes place in a high frequency portion of the spectrum, and which is an example of anticipatory coarticulation (or right-to-left coarticulation). This means that the articulator has moved from the present phoneme (/i/) toward a position (higher frequency) that is more appropriate for the following phoneme (/s/).



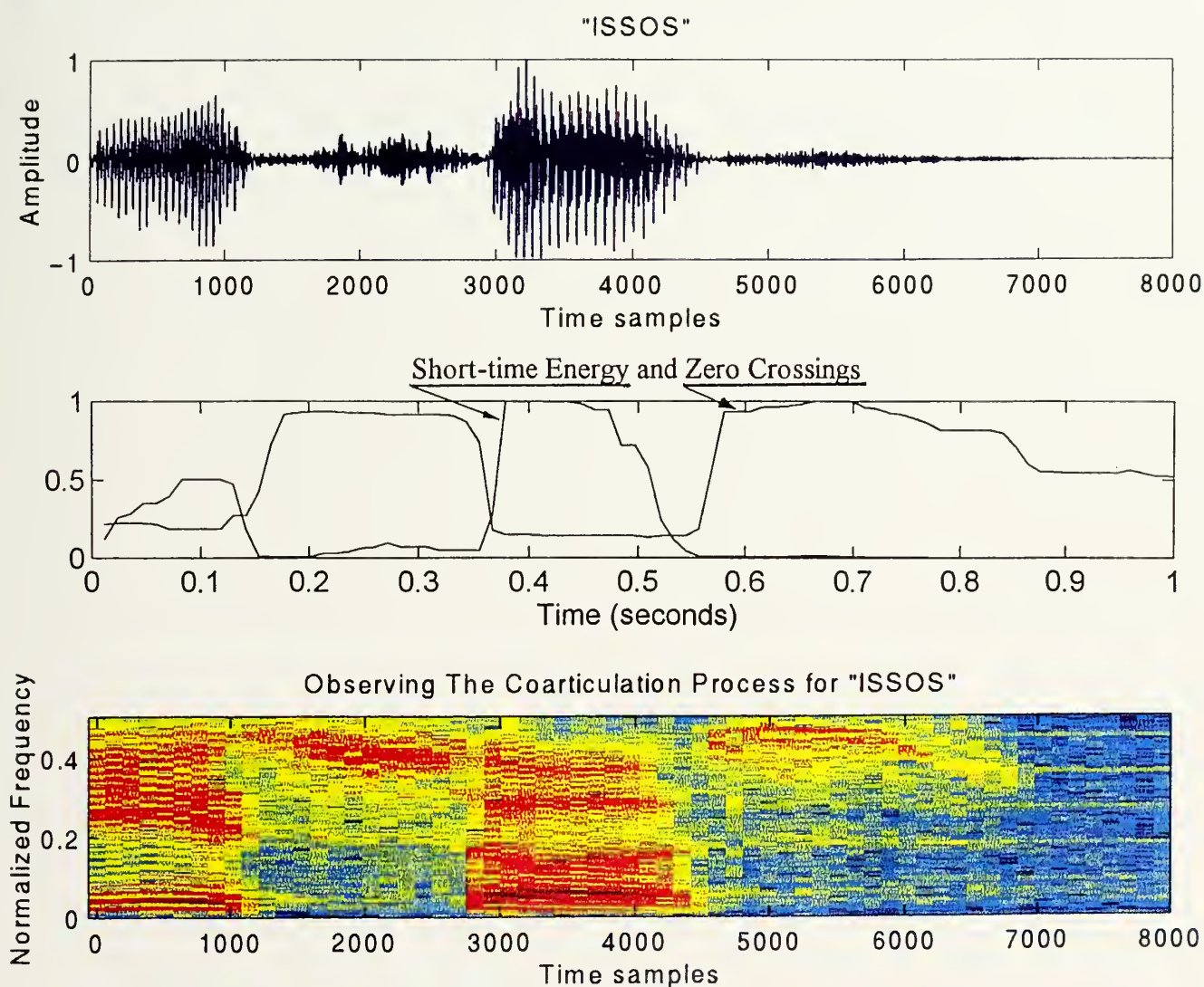


Figure 2.1 Sound "ISSOS," male non-native speaker; top plot: Time domain representation; middle plot: Short time energy and zero-crossing representation; bottom plot: Spectrogram of "ISSOS" using a Hanning time window of length 256 samples with overlap 128,  $f_s = 8$  KHz.









### III. THE LOCAL TRIGONOMETRIC TRANSFORM

This chapter discusses the main concepts related to the Local Trigonometric Transform theory and its implementation. Much of the mathematical rigor is omitted, and emphasis is placed on the basic theory and its application to speech processing. This chapter is divided into six sections. The first provides an introduction, and the second presents some basic definitions about the rising/cutoff function. The third section defines the folding and unfolding operations that are used for the transform [2]. The fourth describes the Continuous Transform and its main mathematical properties. The fifth defines the Discrete Transform. Finally, the last section applies these concepts and describes how the transform may be performed by using orthonormal bases to allow for signal analysis and synthesis.

#### A. INTRODUCTION

In order to analyze small portions of the speech signal, it must be partitioned in time. The local transform defined in this chapter applies a “local cosine,” which is a basis function that allows the signal to be cut into time slices. As first defined by Malvar in 1987 [3], the “local cosines” provided a regularly spaced partition in time. Later, Coifman and Meyer [4] and Meyer [5] tackled the problem of modifying regular constructions to obtain windows with variable lengths that could be defined arbitrarily. They began by partitioning time into adjacent intervals  $[\alpha_j, \alpha_{j+1}]$ , as illustrated in Figure 3.1. Figure 3.2 shows in more detail how the windows may be combined while still preserving the smoothness and integrity of the signal. The windows used are essentially the intervals  $[\alpha_j, \alpha_{j+1}]$ . The disjoint intervals  $[\alpha_j - \epsilon_j, \alpha_j + \epsilon_j]$  allow the windows to overlap. In summary, the local cosines (called “Malvar wavelets”) are constructed with a rising duration ( $2\epsilon_j$ ), a stationary period ( $\Delta t$ ), and a decay (which lasts  $2\epsilon_{j+1}$ ). The ability to arbitrarily and independently choose the duration of the rising and decaying, as well as the stationary section, is exactly what makes the Malvar wavelets different from other well-known wavelets (e.g., Gabor or Daubechies) [5]. Of course, it is important to use this ability

efficiently. This choice will be discussed in the following chapters, where we focus on the best basis for decomposition of the signal.

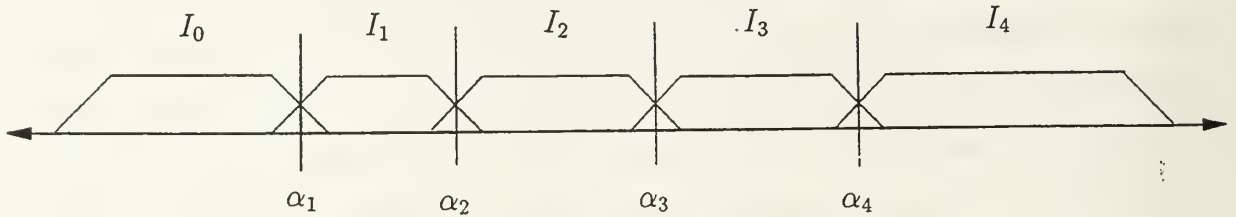


Figure 3.1 Arbitrary Partition of Time into Adjacent Intervals

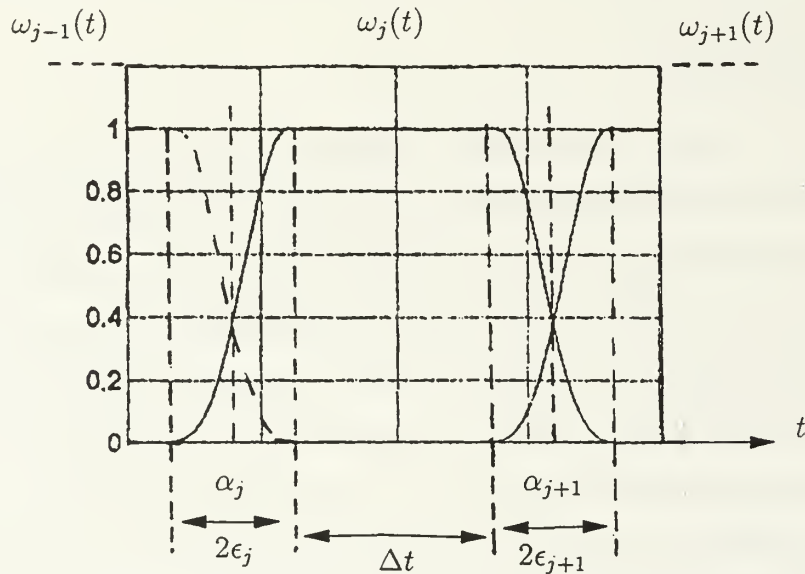


Figure 3.2 Overlapping windows of arbitrary size

## B. THE RISING / CUTOFF FUNCTION

The well-known Discrete Cosine Transform (DCT) has, as its basis function, a “block cosine” (i.e., a rectangular window that multiplies the cosine function). The functions obtained by the block cosine result in a discontinuity or an abrupt variation in

the signal. As a result, we have discontinuities at the block boundaries of the reconstructed signal. The effects produced include the so-called “blocking effect” in image coding, and the “clicking sounds” in speech coding [6]. These problems can be avoided by defining a window based on a function that allows for a smooth transition from zero to the amplitude of the cosine (on the left edge), as well as from that amplitude to zero (on the right edge).

The function  $r$  is defined as  $r = r(t)$  in the class  $C^d(R)$ , for some  $0 \leq d \leq \infty$ , satisfying the following conditions:

$$|r(t)|^2 + |r(-t)|^2 = 1 \text{ for all } t \in R; \quad r(t) = \begin{cases} 0, & \text{if } t \leq -1, \\ 1, & \text{if } t \geq 1. \end{cases} \quad (3.1)$$

It is called a rising cutoff function because  $r(t)$  monotonically increases from zero to one over the domain of  $t$  from  $-\infty$  to  $+\infty$ . That function is presented in Figure 3.3.

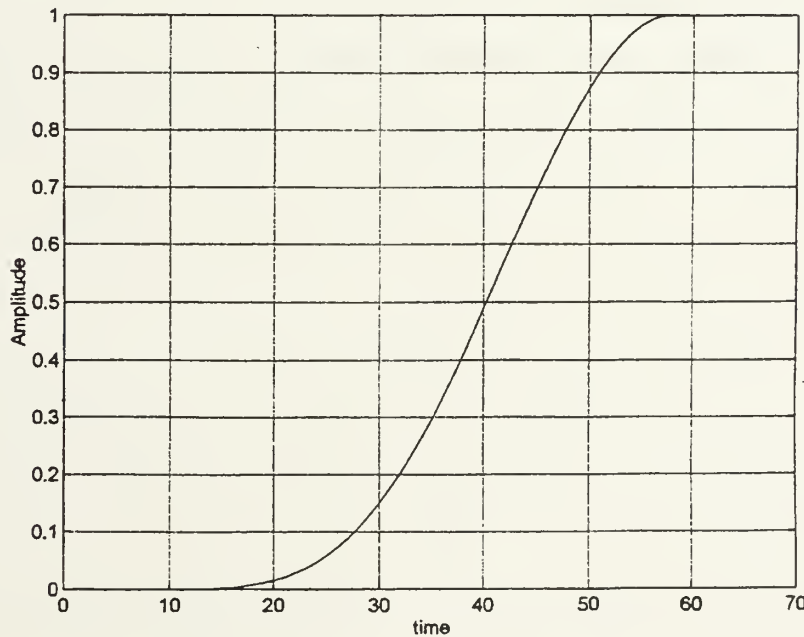


Figure 3.3 The rising cutoff function

### C. FOLDING AND UNFOLDING

The folding operator  $U$  and its adjoint unfolding operator  $U^*$  are defined as follows:

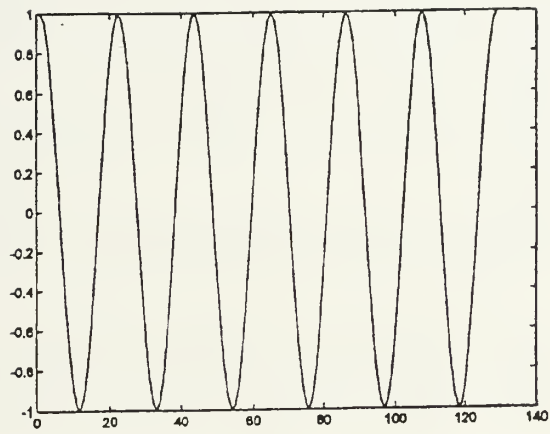
$$Uf(t) = \begin{cases} r(t)f(t) + r(-t)f(-t), & \text{if } t > 0 \\ r(-t)f(t) - r(t)f(-t), & \text{if } t < 0 \end{cases} \quad (3.2)$$

$$U^*f(t) = \begin{cases} r(t)f(t) - r(-t)f(-t), & \text{if } t > 0 \\ r(-t)f(t) + r(t)f(-t), & \text{if } t < 0. \end{cases} \quad (3.3)$$

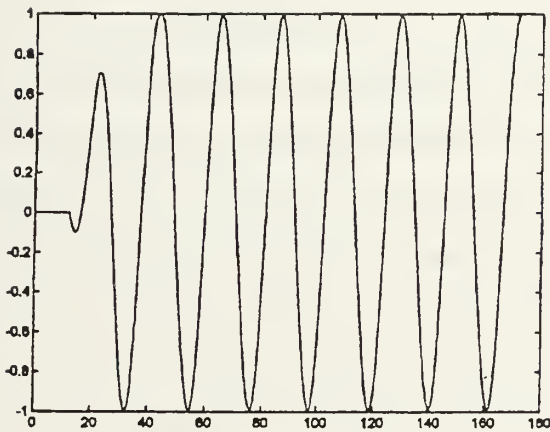
Observe that  $Uf(t) = f(t)$ , and  $U^*f(t) = f(t)$ , if  $t \geq l$  or if  $t \leq -l$ . Also,  $U^*Uf(t) = UU^*f(t) = (|r(t)|^2 + |r(-t)|^2) f(t) = f(t)$ , for all  $t \neq 0$ , so that  $U$  and  $U^*$  are isomorphisms of  $L^2(R)$ . This means that one operator is the inverse of the other.

Figure 3.4 illustrates the unfolding operation on a block cosine. Figure 3.4a shows a block cosine. Figures 3.4b and 3.4c illustrate the cosine unfolded at its left edge, and unfolded at both edges, respectively.

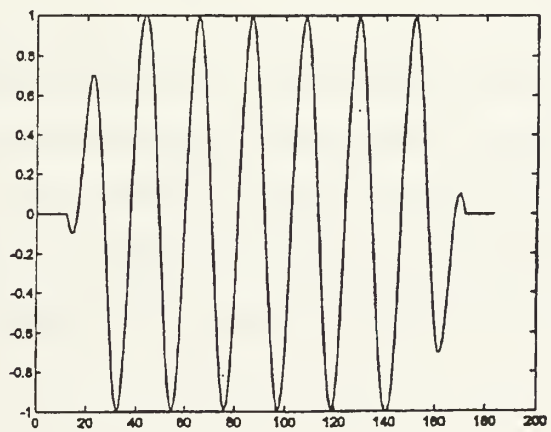
Figure 3.5 presents a block cosine and a block sine after periodic folding. The purpose of folding is to prepare the function intervals, so that the adjacent windows can be overlapped further without changing the function in the overlapping interval.



(a) Block cosine



(b) Left edge unfolded



(c) Both edges unfolded

Figure 3.4 Unfolding operator in a block cosine



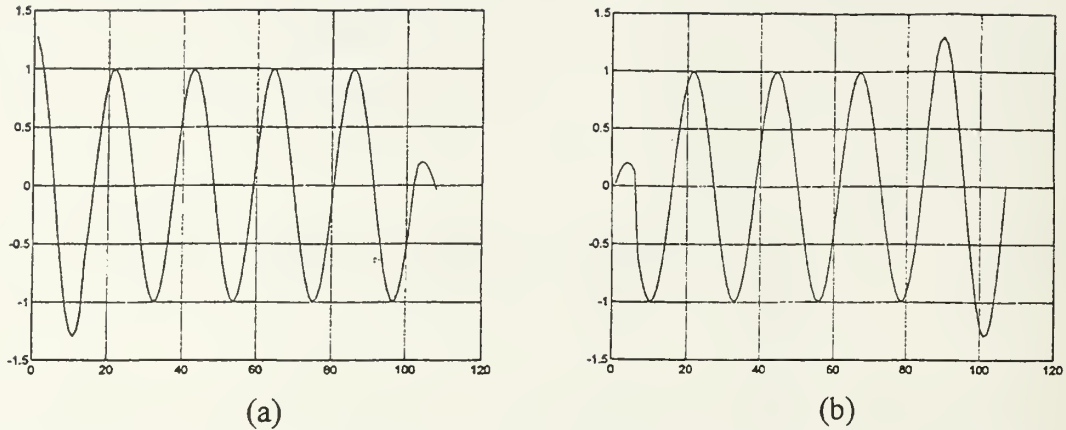


Figure 3.5 (a) Block cosine and (b) Block sine, both after periodic folding

To extend the concept of folding and unfolding in an interval, the operators now can be shifted and dilated so that their action takes place on an arbitrary interval  $(\alpha - \epsilon, \alpha + \epsilon)$ . Now, after partitioning the time by periodically folding the left and right edges of each interval, all the adjacent component windows can be unfolded and overlapped. The window formed by the rising cutoff function is called a bell. Figure 3.6 displays two small bells (called child bells) overlapped and one inverted large bell (called the parent bell) below, showing that it is possible to preserve both the smoothness between intervals and the signal integrity (with no loss of information), if each interval is unfolded and then overlapped. This explains how parent windows may be split into two child windows (in the decomposition phase), and how two child windows may be combined to form one parent window (in the reconstruction phase). This property is particularly important when the concept of the “cosine packets” is introduced.

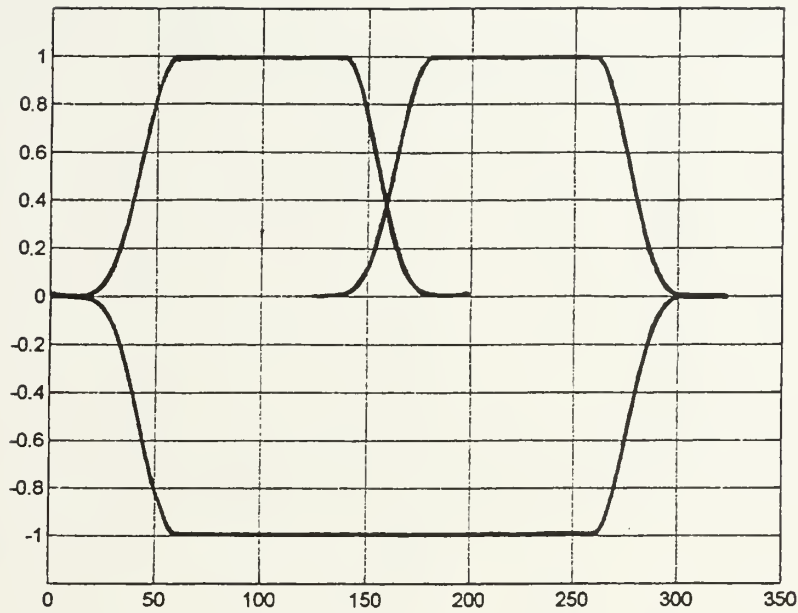


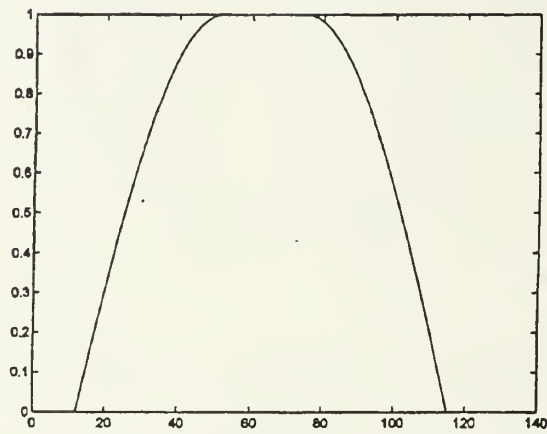
Figure 3.6 Two child bells overlapped and one inverted parent bell

## D. THE CONTINUOUS LOCAL TRIGONOMETRIC TRANSFORM

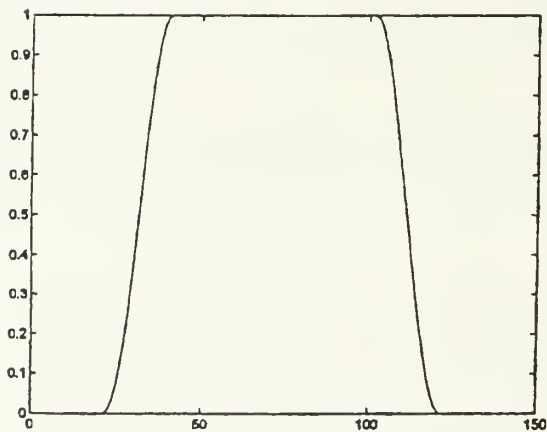
### 1. Properties

The time window used in the Local Trigonometric Transform can have both smoothness and a controlled length so that properties such as time and frequency resolution also may be controlled. This can be implemented simply by changing the equation of the window. By combining windows of arbitrary size (represented by local cosines, i.e., block cosines unfolded at both edges), it is possible to obtain a smooth orthogonal basis. Observe that each window is well localized in time, as well as in frequency. Its temporal support region is the width of that interval given by  $[\alpha_j - \varepsilon_j, \alpha_{j+1} + \varepsilon_{j+1}]$  and, thus, it has position uncertainty at most equal to that width (Figure 3.2). Figure 3.7 presents three different bells, which are called functions  $r_{[1]}$ ,  $r_{[3]}$ , and  $r_{[5]}$ . Figure 3.8

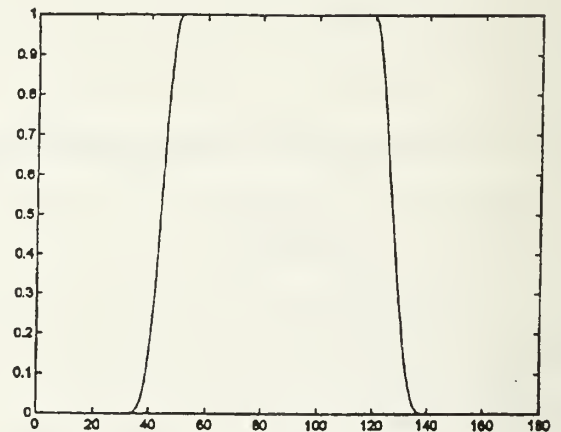
presents the positive half of the real part of the Fourier Transform of the functions given in Figure 3.7.



(a)



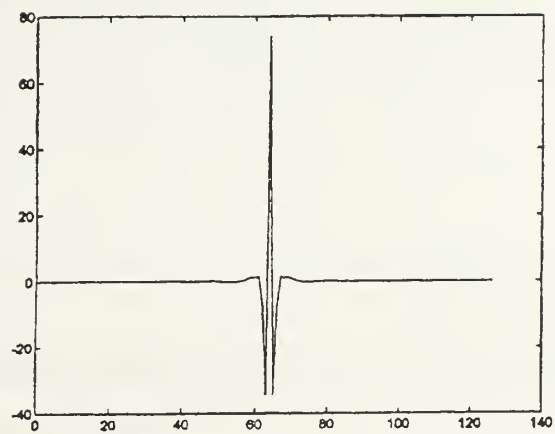
(b)



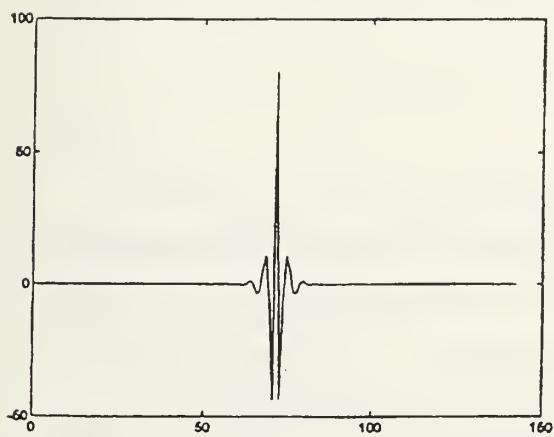
(c)

Figure 3.7 Three different bells; (a)  $r_{[1]}$ ; (b)  $r_{[3]}$ ; (c)  $r_{[5]}$

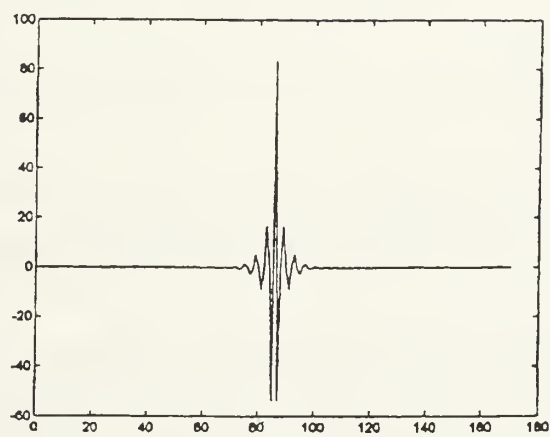
Note that the sidelobes increase as the roll-off of the time window increases.



(a)  $r_{[1]}$



(b)  $r_{[3]}$



(c)  $r_{[5]}$

Figure 3.8 Fourier transforms of the bells of Figure 3.7

## 2. The Local Transform

The Continuous Local Trigonometric Transform is based on a set of orthonormal basis functions that allow for a variable-length time window while still maintaining a small time-frequency bandwidth product. The Transform can be either a local cosine or a local sine. Since the local cosine has been chosen, the definition of the so-called “block cosine” at half-integer frequency is given as follows:

$$C_n(t) = \cos \left[ \pi \left( n + \frac{1}{2} \right) t \right], \quad (3.4)$$

where  $n$  is an integer, and  $t$  is restricted to the interval  $[0,1]$ .

As can be observed from the right side of Figure 3.4c, unfolding the block cosine at the edges gives it the necessary smooth characteristics that contribute to a good frequency resolution for that transform. Basically, smoothness is obtained by a smooth cutoff by sine iteration [2], defined by:

$$\tau_{\sin}(t) = \begin{cases} 0, & \text{if } t \leq -1, \\ \sin \left[ \frac{\pi}{4}(1+t) \right], & \text{if } -1 < t < 1, \text{ and} \\ 1, & \text{if } t \geq 1, \end{cases} \quad (3.5)$$

$$\tau_{[0]} = \tau_{\sin}(t) \quad \text{and} \quad \tau_{[i+1]} = \tau_{[i]} \left( \sin \frac{\pi}{2} t \right). \quad (3.6)$$

Since  $\tau_{[1]}$  is smooth on  $(-1,1)$  with one vanishing first derivative at the boundary points, the envelope (referred to as the bell in [5]) has a continuous derivative on  $R$ . Based on the recursion in Equation (3.6),  $\tau_{[i]}$  can be used with  $i > 1$  to obtain additional derivatives [5]. Actually, it can be shown that  $\tau_{[i]}(t)$  has  $2^{i-1}$  vanishing derivatives.  $\tau_{[1]}$  is used, since it allows good resolution and has very small side lobes.

Thus, the local cosine is defined as :

$$\psi_{n,j} = \sqrt{\frac{2}{\alpha_{j+1} - \alpha_j}} \tau_j \left( \frac{t - \alpha_j}{\epsilon_j} \right) \tau_{j+1} \left( \frac{\alpha_{j+1} - t}{\epsilon_{j+1}} \right) \cos \left[ \frac{\pi \left( n + \frac{1}{2} \right) (t - \alpha_j)}{\alpha_{j+1} - \alpha_j} \right], \quad (3.7)$$

where  $\alpha_j$  and  $\alpha_{j+1}$  are the interval edges,  $\epsilon_j$  and  $\epsilon_{j+1}$  are the action radii of the operators for both edges, and  $r_j, r_{j+1}$  is the rising function  $r_{[1]}$ , applied at both edges of the interval. Note that the local cosine as defined by Equation (3.7) is the result of the unfolding operation at both edges of the “block cosine,” i.e.,

$$\Psi_{n,j}(t) = U^*(r_{j1} \alpha_{j1} \epsilon_j) U^*(r_{j+1} \alpha_{j+1} \epsilon_{j+1}) \cdot 1_{IJ}(t) C_{n,j}(t),$$

where:

- $1_{IJ}(t) C_{n,j}(t)$  represents the block cosine function for an interval beginning at edge  $j$ ;

- $U^*(.)$  is the unfolding operator applied at the left ( $j$ ) and right edge ( $j+1$ ) of the interval.

Thus, the Continuous Local Trigonometric Transform is the inner product  $\langle f, \psi_{n,j} \rangle$ , where  $\psi_{n,j}$  is the local cosine defined above.

Instead of computing in that manner, one may fold the function first, and then obtain the inner product with the regular “block cosine,” as in the expression below:

$$\langle f, \psi_{n,j} \rangle = \langle U_j U_{j+1} f, 1_{IJ} C_{n,j} \rangle. \quad (3.8)$$

In practice, this simple observation has great importance, since it means that  $f$  can be preprocessed by folding, and the local cosine transform can be computed with an ordinary cosine transform [2].

It is also important to observe that, by defining the transformation as an inner product, what is measured is the amount of “similarity” between the signal  $f(t)$  and the basis function  $C_{n,j}$ . This is one of the key attributes that make the local cosine transform convenient for the transformation of speech signals and, therefore, good for compression and coding. The fact that speech can be considered a locally stationary signal with a

reasonable correlation to sines and cosines may explain some of the good results when using a Local Trigonometric Transform.

## E. THE DISCRETE COSINE TRANSFORM

By replacing just the variables with integers, and by using the discrete cosine transform, it is possible to obtain discrete versions of the local cosine. So Equation (3.9) is exactly the same formula as Equation (3.7), but with the variables replaced by integer values. In Equation (3.9) it is assumed that:

- $\alpha_j < \alpha_{j+1}$ , where  $\alpha_j$  and  $\alpha_{j+1}$  are integers;
- the signal is sampled at integer points  $t$ ,  $\alpha_j \leq t < \alpha_{j+1}$ , which gives  $(\alpha_{j+1} - \alpha_j)$  samples;
- $r_j$  and  $r_{j+1}$  are the rising functions  $r_{[1]}$ , applied at both edges of the interval;
- $\epsilon_j > 0$  and  $\epsilon_{j+1} > 0$ , with  $\epsilon_j + \epsilon_{j+1} \leq$  number of samples to insure that the action regions are disjoint.

Equation (3.9) also makes a distinction between the left and right endpoints, because sampling is done at the left endpoint of each interval. If sampling is done in the middle of the intervals (which can be done by taking the function in Equation (3.7) and replacing every instance of  $t$  with  $t+1/2$ ), it will be more symmetric, and the basis functions will be cosines sampled between grid points. The result is the following discrete local cosine basis function:

$$\psi_{n_j \text{ (DCT-IV)}}(t) = r_j \left( \frac{t + \frac{1}{2} - \alpha_j}{\epsilon_j} \right) r_{j+1} \left( \frac{\alpha_{j+1} - t - \frac{1}{2}}{\epsilon_{j+1}} \right) \sqrt{\frac{2}{\alpha_{j+1} - \alpha_j}} \cos \left[ \frac{\pi(t + \frac{1}{2})(t + \frac{1}{2} - \alpha_j)}{\alpha_{j+1} - \alpha_j} \right] \quad (3.9)$$



## F. APPLICATION TO SIGNAL ANALYSIS/SYNTHESIS

Given an arbitrary partitioning of a signal in time, it is possible to construct several smooth orthogonal bases, using the local cosine transform as the basis function. The scheme that leads to the best partition and the best basis for this application will be introduced in the next chapter. This section explains how the DCT-IV can be used for an analysis in the frequency domain and for further synthesis in the time domain.

As mentioned in sections “C” and “D”, the signal is first folded at the left and right ends of each interval. Then, an ordinary DCT-IV transform is used to compute the Local Cosine Transform for each of the windows obtained. Now, it becomes possible to analyze each time window using the frequency spectrum (from DC to  $f_s/2$ , where  $f_s$  is the sampling frequency). To reconstruct the signal, the DCT-IV is applied to obtain the inverse. As in the decomposition phase, the transform is computed first with the regular “block cosine,” and then the intervals are unfolded, instead of using the local cosine. By periodically unfolding the left edges of the current interval and the right edge of the following one, the smoothness and integrity of the function are preserved, allowing the time domain function to be reconstructed.



## IV. WAVELET AND COSINE PACKET TRANSFORMS

This chapter presents the Wavelet Transform and two general time-frequency analysis schemes: the Wavelet Packet Transform and the Cosine Packet Transform .

### A. INTRODUCTION

The goal of this thesis is to obtain the scheme best suited for the decomposition and reconstruction of speech signals, in particular, one that can decompose a speech signal into an orthonormal basis function. First, the Wavelet Transform (WT) and its main properties and characteristics are discussed. Next, the general concept of the Wavelet Packet Transform (WPT) is introduced. Finally, the Cosine Packet Transform (CPT) is presented. This last scheme initially performs a time split, as opposed to transforms that first split the signal in the frequency domain.

### B. THE WAVELET TRANSFORM

In the Wavelet Transform (WT) algorithm, the sampled data set is passed through the low-pass and high-pass filters with complementary bandwidths, known as quadrature mirror filter (QMF) pairs [7]. The outputs of both filters are decimated by a factor of two. So, at each scale, we have a set of high-pass filtered data and a set of low-pass filtered data. Each of these sets has half as many elements as the original data set, as a consequence of the decimation. The low-pass filtered data can be used as the data input for another pair of filters identical to the first pair, generating another set of low- and high-pass coefficients at the next lower level of scale [8].

This process can continue until the set of original coefficients has been reduced to the minimal scale level, which is two coefficients. Figure 4.1 presents the pyramid algorithm of the WT. Figure 4.2 shows how a unit interval of length  $2^j$  samples can be decomposed to obtain a maximum of  $j$  levels of transform data. Figure 4.3 presents the tiling diagram that corresponds to the WT decomposition. This shows that the WT works well if the signal is composed of strong components of short duration, i.e., bursts. This

means that the WT is a good detector of transients. It also works well if the signal is composed of low-frequency components of long duration [9].

As stated earlier, speech is composed of portions of either high frequency or low frequency, both with a typical minimum duration of about 15 milliseconds. These characteristics indicate that the WT may not be the best scheme for speech signal analysis.

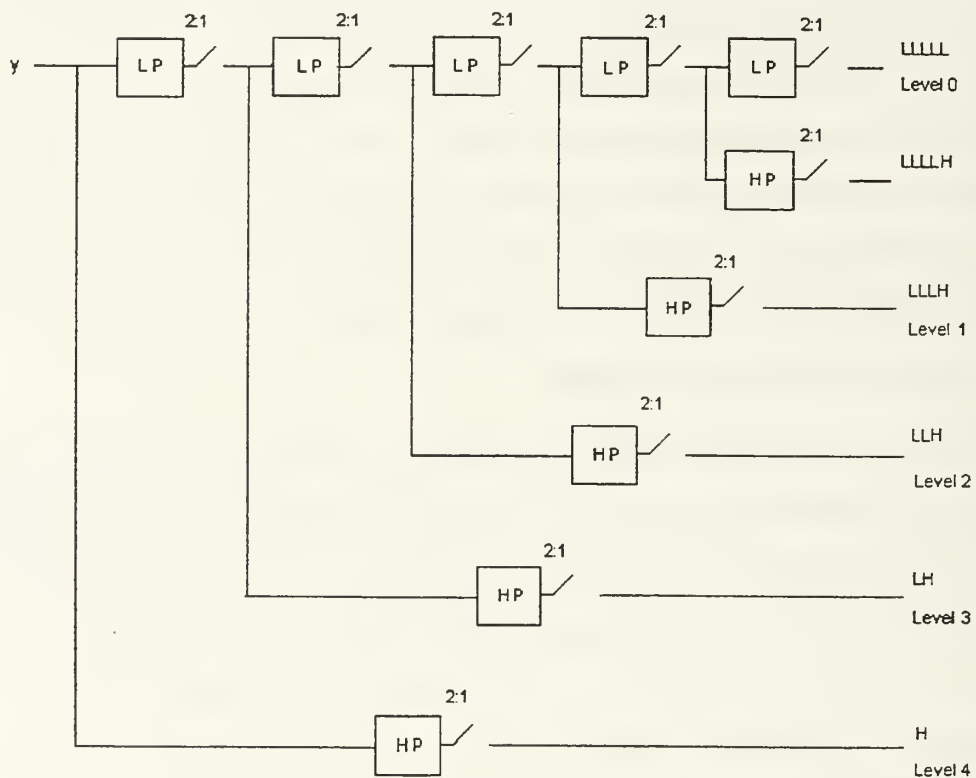


Figure 4.1 WT implementation: A bank of QMF pairs

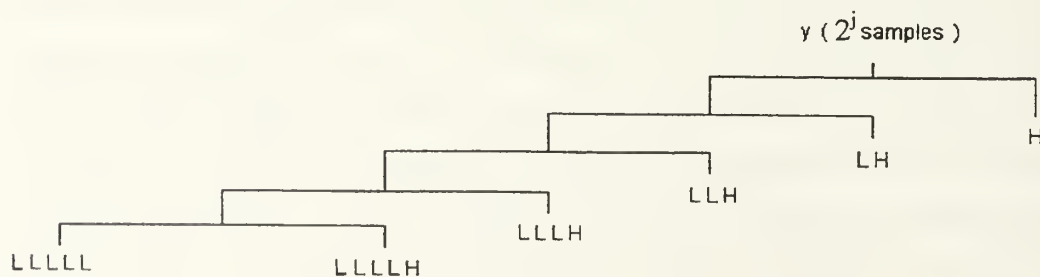


Figure 4.2 Wavelet transform: decomposing  $2^j$  samples into a maximum of  $j$  levels

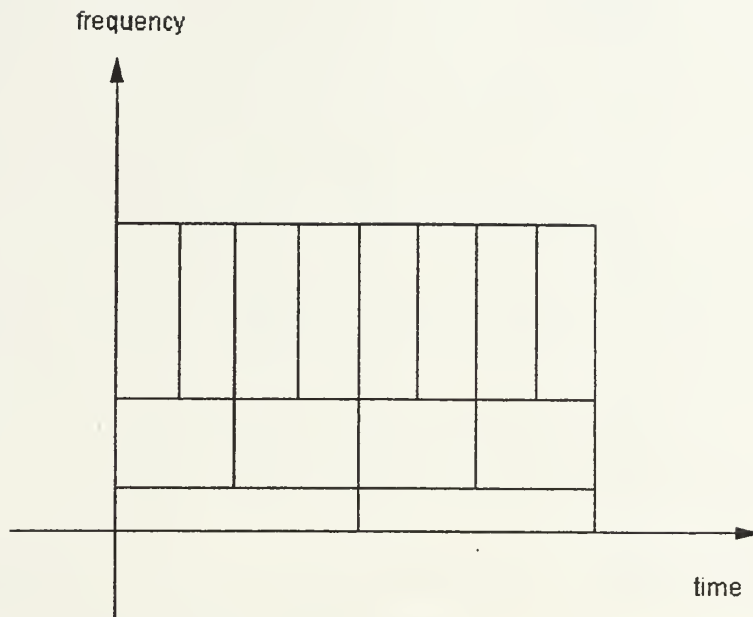


Figure 4.3 WT tiling diagram

### C. THE WAVELET PACKET TRANSFORM

The WT is not the only way to split the signal in the frequency domain. The Short Time Fourier Transform (STFT), for example, is another possible scheme. However, in the STFT, both the time and frequency resolution are kept constant by the choice of the time window length (Figure 4.4).

Actually, both the WT and the STFT can be viewed as part of a general scheme called the Wavelet Packet Transform (WPT), which is a collection of possible sets of orthonormal basis functions.

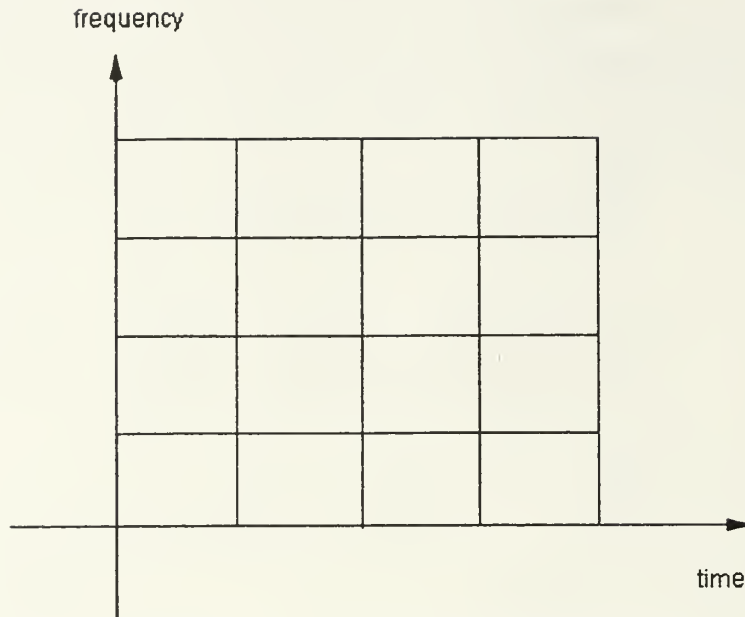


Figure 4.4 Tiling diagram for the STFT

Figure 4.5 depicts the general tree structure for the WPT. Note that the heavy lines indicate the graph that forms the WPT basis. The symbol L or H has been assigned to each half frequency division, depending on whether it is a high- or low-frequency band. Following the tree structure, we have assigned those symbols sequentially, following the same rule. Note that the WT basis consists of the subspaces H, LH, LLH, LLLH and LLLL.

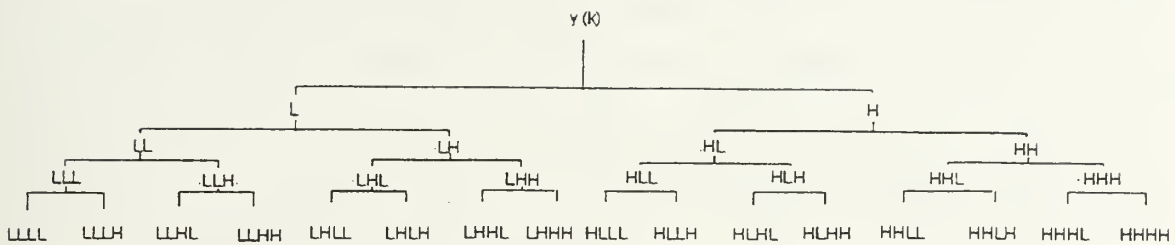
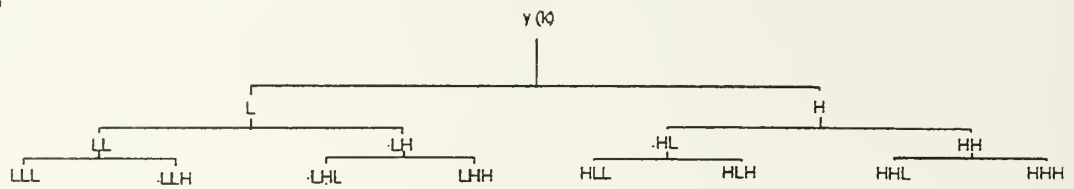


Figure 4.5 General tree structure for the WPT

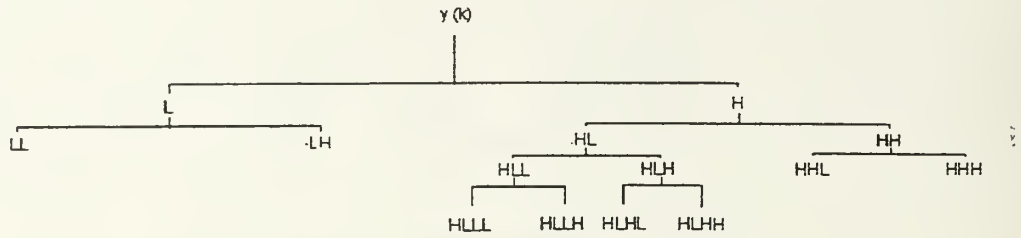
The sequences L, LL and LLL are intermediate steps leading to the generation of the subspaces of the wavelet basis at the lower levels.

Since the frequency splitting results in the low- or high-pass version of the filtered data (i.e., either half branches of the tree),  $j2^j$  graphs representing different orthonormal bases can be created. Figure 4.6 presents three different Wavelet Packet decompositions. The basis is a subband decomposition scheme [10], where the basis obtained is composed of the eight bottom divisions. The second is another possible decomposition leading to an orthonormal basis. The third decomposition is exactly the opposite of that obtained using the WT. Figure 4.7 illustrates the tiling diagram that corresponds to the third decomposition. Note the higher frequency resolution for higher frequencies, and the higher time resolution for lower frequencies.

(a)



(b)



(c)

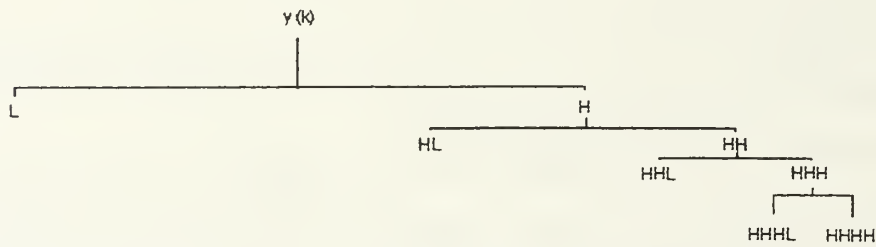


Figure 4.6 Three different wavelet packet decompositions leading to three different bases

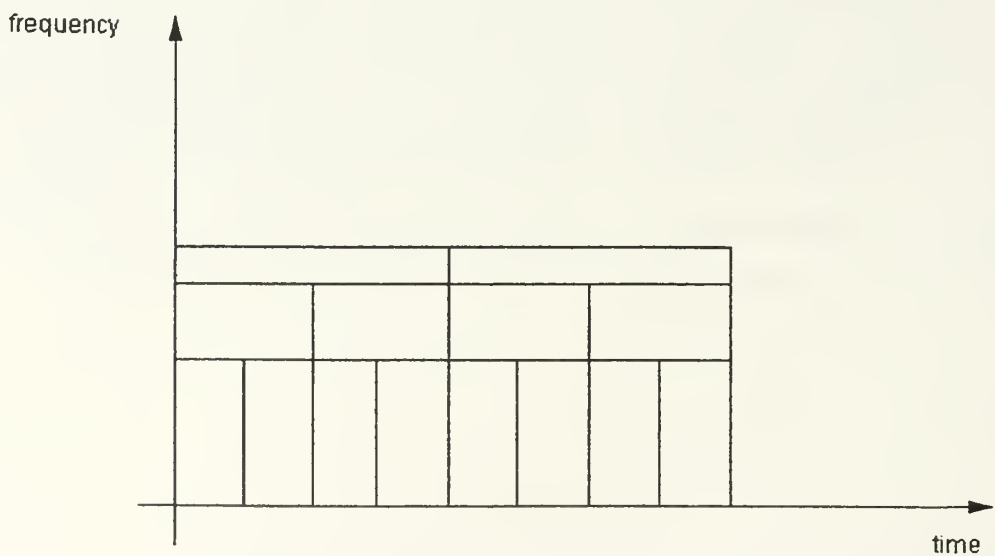


Figure 4.7 Tiling diagram for the decomposition of figure 4.6c



#### D. THE COSINE PACKET TRANSFORM

The Cosine Packet Transform (CPT) is a scheme that allows for a time-splitting decomposition prior to the frequency transformation. If one imagines the original signal in the time domain being split successively into two halves at each iteration, a tree configuration will result (Figure 4.8). If the transform imposes no restriction on the support intervals of the window envelopes, the tree does not need to be homogeneous. This means that the windows do not need to be combined in the same way (either in pairs or in any other specific manner). Also, the subspaces do not need to be of equal size. So, in analogy to the wavelet packets case, one is now faced with a large number of possible orthonormal basis configurations, each one of them being considered as a cosine packet. It is important to observe that in the cosine packets case, the windows do not need to be of a dyadic size, they may be of an arbitrary size. However, in this thesis, only dyadic sized windows are considered.

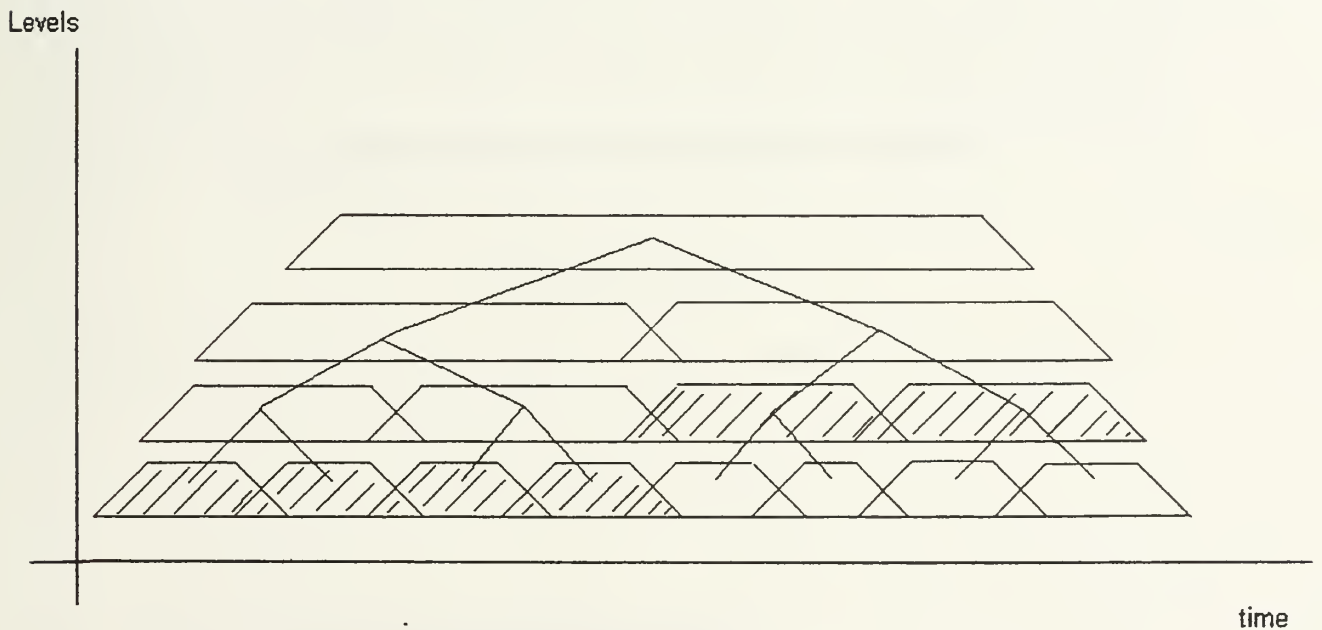


Figure 4.8 Cosine packet transform: The tree configuration

We also note that, as one goes down the tree, time resolution is improved by a factor of two at each layer, while frequency resolution is decreased by a factor of two at

each iteration. Figure 4.9 presents the tiling diagram that corresponds to the tree configuration shown in Figure 4.8. The CPT works in such a way that, after time splitting to a certain depth, a basis is selected by some criterion. Then, for each time window, the DCT-IV transform is applied.

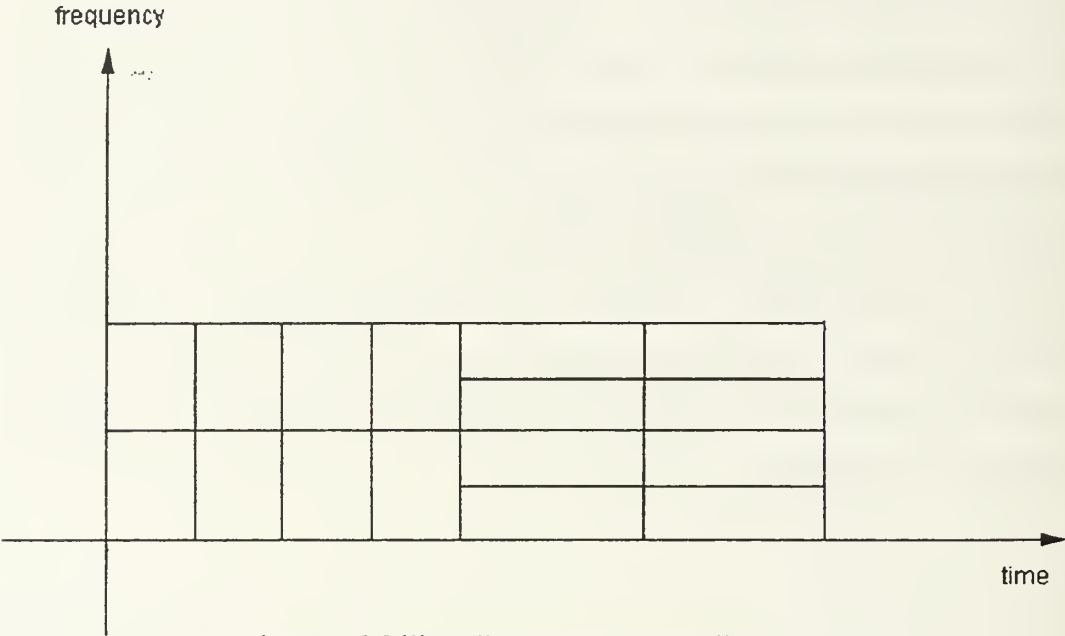


Figure 4.9 Tiling diagram corresponding to Figure 4.8

## V. THE BEST BASIS ALGORITHM

### A. INTRODUCTION

When a choice of bases exists for the representation of a signal, it is possible to determine the best one using some predetermined criterion. The criterion will always depend on the type of signal and the user's objective. In this case, the signal is speech and the objective is to minimize the number of symbols used to represent the information contained in a given interval (i.e., it is desirable to minimize the entropy of that interval). The "best basis" criterion allows for the minimization of some information costs options, including the entropy minimization method [6,11].

We recall that the entropy of a vector  $u = \{ u(k) \}$  is defined by :

$$H(u) = \sum_k p(k) \log(1/p(k)), \quad (5.1)$$

where  $p(k) = |u(k)|^2 / \|u\|^2$  is a normalized energy of the  $k^{\text{th}}$  element of the sequence, and  $p \log 1/p$  is set to 0, if  $p = 0$ .  $H(u)$  is the entropy of the probability distribution function (or pdf) given by  $p$ . Note that  $H(u)$  is not an information cost functional, i.e., it is not a direct function of the sequence  $\{u(k)\}$ . But the functional

$$l(u) = \sum_k |u(k)|^2 \log(1/|u(k)|^2)$$

is a direct function. If  $l(u)$  is minimized, then  $H(u)$  is also minimized in the expression:

$$H(u) = \|u\|^{-2} l(u) + \log \|u\|^2. \quad (5.2)$$

### B. THE BEST BASIS ALGORITHM METHOD

Initially, the algorithm computes the entropy obtained in all intervals or "nodes" of the tree. Figure 5.1 presents an example of the cosine packet tree with corresponding computed entropies. The Best Basis Algorithm searches the tree in a bottom-up direction

and, whenever a parent node has a lower cost than that of its children, the Best Basis algorithm flags the parent. If the sum of the children's costs is lower than that of the parent node, this lower cost is assigned to the parent. Similarly, children are flagged when they have a lower information cost than their parents. This step avoids the need to examine any node more than twice: once as a child and once as a parent. Figure 5.2 presents the new and the former (in parenthesis) information costs for each node shown in Figure 5.1. Then, after all nodes present in the tree have been examined, the Best Basis Algorithm selects the topmost flagged nodes, which constitute a basis. Finally, as the topmost flagged node is encountered, the remaining nodes in the corresponding subtree are discarded. Figure 5.3 displays the best basis nodes for this example as shaded blocks. Further details may be found in [4]. Figure 5.4 shows a Best Basis tiling scheme resulting from the decomposition shown in Figure 5.3. It is obvious that each resulting cell occupies one portion of the time, and the whole frequency spectrum is covered by each of those cells.

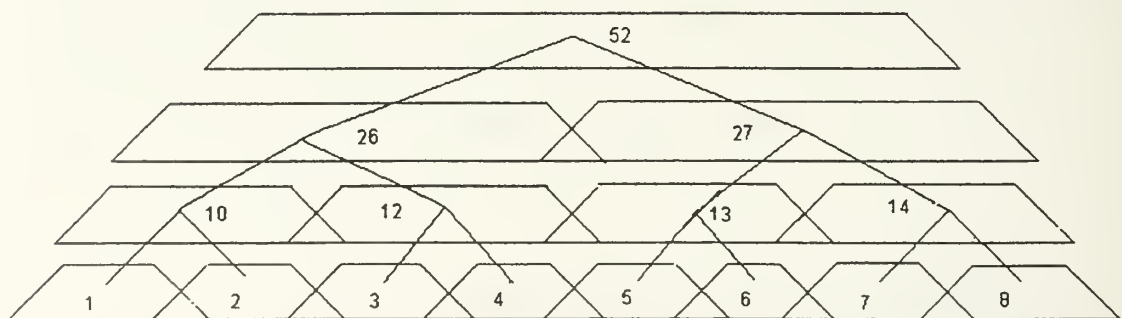


Figure 5.1 Cosine packet tree with computed entropies for every interval (node)

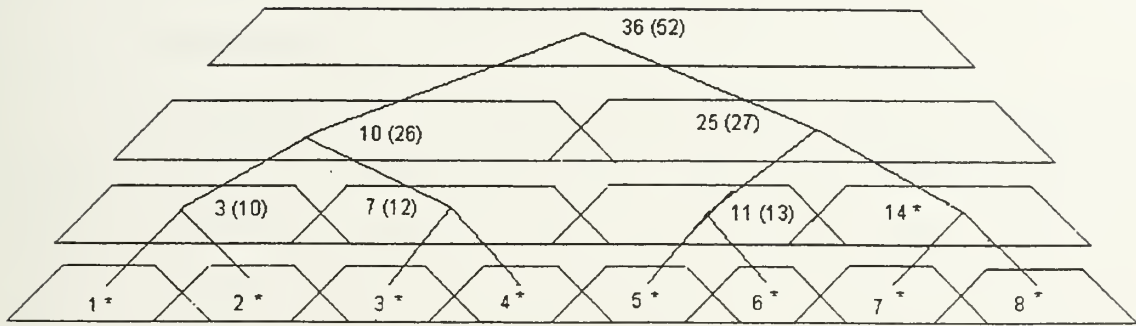


Figure 5.2 New (and former) computed entropy for each node

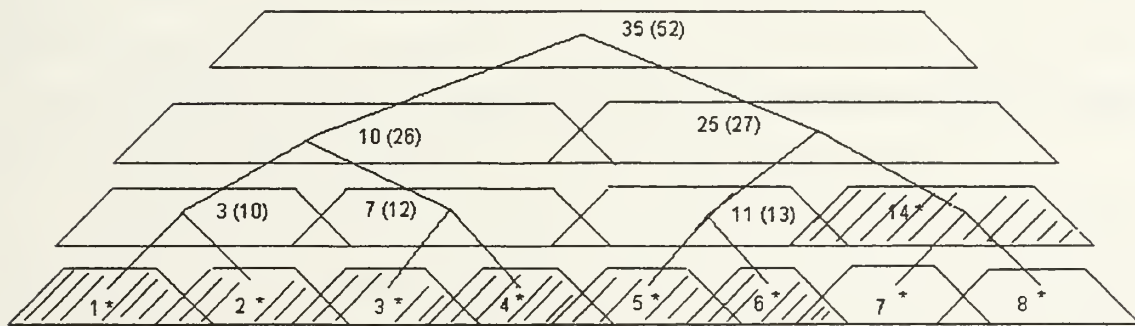


Figure 5.3 Selection of minimum entropy basis

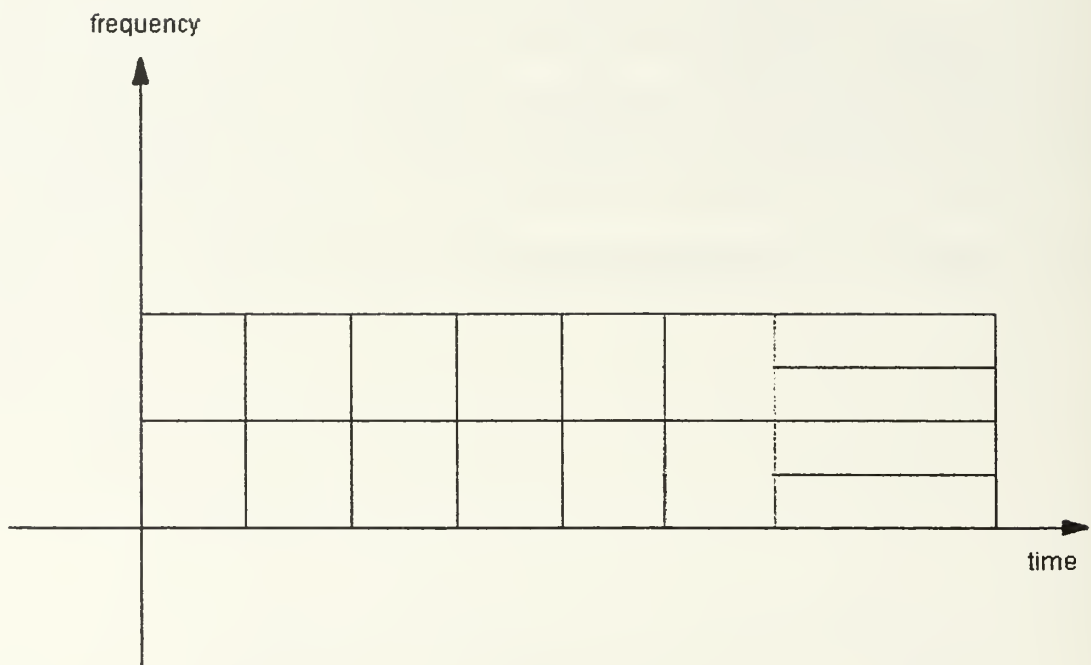


Figure 5.4 Best basis tiling scheme resulting from the decomposition in Figure 5.3

## VI. COMPRESSION AND DENOISING SCHEMES

This chapter describes the compression and denoising schemes used in this research. It is divided into five sections. First, the motivating concepts are introduced. In the remaining sections: minimum time window, voiced-unvoiced segmentation, adaptive thresholding and denoising are presented.

### A. INTRODUCTION

Initial research for this thesis included reviewing existing lossy compression techniques, which are divided into two main classes: Lossy Predictive Coding and Transform Coding [12]. The attention of this thesis is directed to Transform Coding. The Transform Coding technique that has been largely discussed, applied, and tested is the Wavelet Transform. However, as explained in Chapter IV, wavelets are more appropriate for the analysis of either transients or long-duration, low-frequency stationary signals than for speech signals.

As shown in Chapter III, the Local Cosine Transform has good time and frequency resolution. Also, unlike the Fourier Transform, the Discrete Cosine Transform IV (DCT-IV) decorrelates the signal in each window, which facilitates compression. Experiments for this research demonstrated that the Best Basis Algorithm, besides selecting the basis with minimal entropy, is also able to split the speech signal into locally stationary time segments. As a result, the combination of the Cosine Packet scheme with a method that selects the Best Basis (BB) configuration to minimize the entropy in each interval seems to be most appropriate for the applications considered here.

An important characteristic of the Cosine Packet Transform (CPT) is that it allows time resolution to be controlled. If one uses the WPT with the Best Basis Algorithm on speech, the algorithm chooses the basis based on the minimization of some information cost of the frequency coefficients. Thus, in the WPT case, time resolution is not a function of the physical properties of speech. Instead, it is dependent on each scale which in turn is selected by the best basis criterion. Also, the user must select the maximum



frequency splitting depth by choosing the worst (largest) time resolution, not the best. With the CPT, on the other hand, it is possible to choose the depth and, thus, to determine the minimum time interval, which ideally should coincide with the minimum locally stationary portions of speech.

Once the signal is divided into its locally stationary intervals, the DCT-IV algorithm is applied to transform the signal to the frequency domain. Then, for all of the time windows, the signal is passed through a thresholding scheme that picks up different percentages of coefficients, according to the frequency and energy contents of each frame. Basically, the speech is divided into its voiced and unvoiced sounds, making it necessary to implement a scheme for voiced-unvoiced segmentation.

Recordings made for this research included noise generated by the equipment. This noise was composed basically a of 60 Hz hum and harmonic components. Since the noise frequencies in each time window were detectable, it was possible to denoise the words and sentences used in the experiments. The system is composed of the three main blocks, as shown in Figure 6.1.

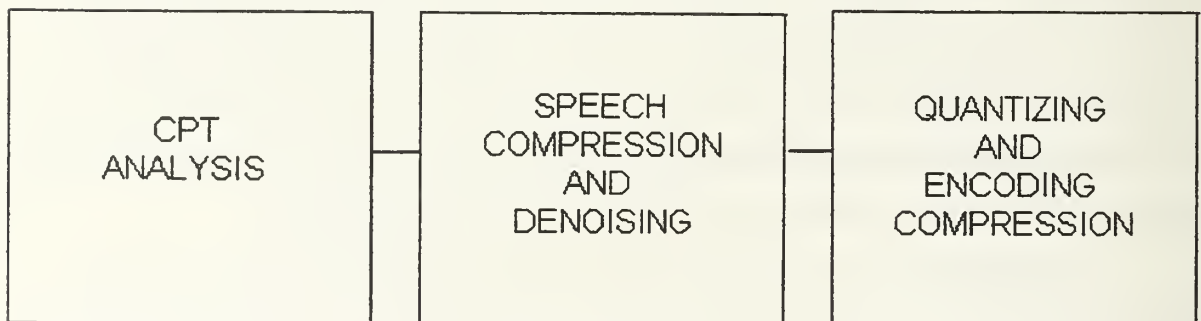


Figure 6.1 System block diagram

The Cosine Packet scheme, presented in Chapters IV and V, is based on the CPT and Best Basis Algorithm. The encoding/compression schemes investigated in this work will be presented in Chapter VII.

## **B. MINIMUM TIME WINDOW SIZE**

The choice of the minimum time window depends on the time and frequency resolution desired. We recall that, in the CPT scheme, the further down on the tree, the better the time resolution, and the worse the frequency resolution. A second consideration is to represent a clean signal in an optimal way, so that the DCT-IV coefficients (in the frequency domain) lead to the smallest number that best represent the energy and frequency content of each interval. Ideally, the signal should be divided into the exact locally stationary portions of the speech, each beginning and ending at the correct points. This is to obtain good compression ratios, where each time interval should have one or two representative coefficients.

The best minimum window sizes were 32 or 16 milliseconds for most of the experiments, and 8 milliseconds for some of them. Since samples were taken at 8 KHz, this means that the intervals are 256, 128, or 64 samples, respectively. Using windows shorter than 16 ms degraded the frequency resolution for most of the test words and sentences, which led to the following two results:

- (1) Loss of coarticulation;
- (2) Degradation in denoising performances.

Although the depth corresponding to the 16-ms minimum-size window was not always the one that gave the best (least) mean square error ( i.e., comparing to 32-ms and 8-ms test windows), the difference obtained in that parameter was not large enough to justify choosing another depth. This was mainly due to the quality factor in reconstruction. Consequently, 16 milliseconds was selected as a compromise for the minimum window size.

### C. VOICED-UNVOICED SEGMENTATION

This section presents an experiment based on the voiced-unvoiced segmentation scheme proposed by Wesfreid and Wickerhauser [13]. Recognition of certain excitation types was attempted to obtain the best possible scheme for compression. Therefore, speech partitioning became one of the subproducts of this research. Once each interval's magnitude spectra and energy are obtained, it is possible to identify voiced and unvoiced portions of the speech.

The spectrum is divided into six main frequency ranges. Table 6.1 displays the low and high frequencies in each range, as well as the corresponding amplitudes of the vertical bars used to separate the intervals.

<i>Frequency Range(Hz)</i>		<i>Vertical Bars</i>
<i>Low</i>	<i>High</i>	<i>Amplitude</i>
0	250	0.1
251	500	0.25
501	1,000	0.5
1,001	2,000	1.0
2,001	3,000	2.0
3,001	4,000	2.5

Table 6.1 Frequency ranges and display

Figure 6.2 illustrates the short-time energy and zero-crossing plots (top) from Voicedit, from the SPC Toolbox [16], for the sentence “/This place blows/” (bottom).

Figure 6.3 presents four plots. The first shows the time domain plot. The second and third plots show, respectively, the frequency behavior according to Table 6.1, and the energy behavior obtained by summing the squares of the coefficients in each interval. The fourth plot (bottom) shows the spectrogram of the speech signal. Note that the tendency of both frequency and energy plots match those of Figure 6.2.

Voiced-unvoiced segmentation obtained the best results when all the intervals with the largest coefficient positioned at a frequency below 1,000 Hz, and energy above a certain threshold were assigned as voiced. All the intervals with the largest coefficient at a frequency above 1,000 Hz were assigned as unvoiced. Figure 6.3 illustrates that a voiced sound results in a high energy and low frequency (largest coefficient frequency below 1,000 Hz) representation for those segments. This is the case for the sounds “/i/,” “/a/,” and “/o/.” In turn, unvoiced sounds are recognized as segments with high frequency (largest coefficient frequency above 1,000 Hz) and low energy content. This is the case of the sounds “/s/” from “this” and “place.” Figure 6.4 shows the result of the voiced-unvoiced segmentation scheme, which can be observed in the middle plot. The bottom plot contains the corresponding spectrogram. Figures 6.5, 6.6, and 6.7 present the same kind of plots for the sentence “Be nice to your sister.” Again, the voiced sounds “/aI/,” “/o/,” and “/i/” are distinguishable from the unvoiced “/s/,” and “/t/.”

#### **D. ADAPTIVE THRESHOLDING**

This section utilizes the partitioning of speech into voiced-unvoiced segments to implement an adaptive scheme for selecting cosine packet coefficients.

Experiments showed that a more natural sounding speech was reconstructed after compression when using more coefficients to represent voiced than unvoiced segments. This resulted in the use of a different percentage of coefficients in the following four cases:

- A) Low frequencies, low energy

- B) Low frequencies, high energy
- C) High frequencies, low energy
- D) High frequencies, high energy.

The need to select more coefficients to represent the voiced segments of speech is illustrated in the example where the isolated noise-free word “nice” is compressed. As explained in Section B, the minimum window size chosen is 16 ms. Figure 6.8 shows that, when the compression scheme is set to keep one cosine packet coefficient per 16 ms window to represent the phoneme /i/, the higher formants of that phoneme are lost. As a result, the phoneme /i/ tends to sound like a /u/. This example illustrates the fact that more than one coefficient may be required to represent voiced phonemes accurately. Figure 6.9 presents the plots that result when two CP coefficients per 16 ms window are selected to represent voiced phonemes (including phoneme /i/), and one CP coefficient out of every 16 or 32 ms interval is selected to represent unvoiced phonemes. Although a lower mean value is achieved for the percentage of selection (and, thus, a higher compression rate), the sound /i/ is correctly reconstructed without affecting the other phonemes of the word “nice.”

Similar findings were obtained with other voiced phonemes such as /a/ and /o/. In addition, experiments showed that the voiced plosive /p/ was degraded by the compression process and sounded like a /b/. Keeping three cosine packet coefficients per 16 ms window interval for voiced segments led to a more accurate representation of the information after compression, as confirmed by the smaller MSE and better sound quality in the reconstructed signal. Further experiments showed that one cosine packet coefficient per 16 ms interval is sufficient to represent the unvoiced segments accurately.

## E. DENOISING

Previous sections have considered only the problem of compressing noise-free signals. However, some of our recordings had a significant amount of low frequency



equipment noise located around 60 Hz and some of its harmonics. As a result, a denoising step was investigated prior to compressing the data to improve the quality of the compressed signal. Thus, the noisy speech signal was denoised prior to applying the compression scheme. The denoising code is given in the Appendix.

Two different cases where noise was present were considered: *Noise-only data segments* and *noisy speech segments*. Noise-only data segments occur before and after isolated word recordings, and between words in the sentence recordings. Experiments showed that the cosine packet coefficients allowed the detection of noise-only segments. The following two situations characterizes the noise-only case according to implementation `ndencomp.m`, given in the Appendix:

(1) Whenever the largest coefficient in the segment is at a frequency less than or equal to 62.5 Hz, and the second largest coefficient is at a frequency less than or equal to 300 Hz or higher than 1,000 Hz;

(2) Whenever the largest coefficient in the segment is in a frequency range between 62.5 Hz and 250 Hz, and the second coefficient is at a frequency less than 200 Hz.

The following situations characterizes the noise-only case according to implementation `encp6.m`, given in the Appendix:

(1) The largest coefficient in the segment is at a frequency less than or equal to 125 Hz, and the second largest coefficient is at a frequency less than 300 Hz;

(2) The largest coefficient is at a frequency less than 62.5 Hz, and the second coefficient is at a frequency higher than 1,000 Hz;

(3) The largest coefficient is at a frequency less than 500 Hz for the female speaker, or less than 1,000 Hz for the male speaker, and the second coefficient is at a frequency less than 125 Hz.

All remaining cases are considered as noisy speech. For those cases, all coefficients located at frequencies below or equal to 62.5 Hz are zeroed out.

Three specific noise-and-speech cases are presented as follows:

(1) *Noisy speech—Case 1.* An example of this case is the word “hey,” where the sound /h/ was lost in the background noise. Due to the higher frequency content of “/h/” (as opposed to the noise), it was possible to identify and pick up one more CP coefficient per interval; thus, retrieving the sound of “/h/.” This example is illustrated in Figures 6.10 and 6.11, which show time plots and spectrograms that correspond to keeping 1 CP and 2 CP coefficients/16- ms interval, respectively.

(2) *Noisy speech—Case 2.* This problem required differentiation of the noise-only case from the noisy voiced stops /b/ and /p/. Distinguishing these sounds from noise was easier than case 1 above, since the first largest coefficient obtained for those two phonemes was never less than 250 Hz, making it possible to denoise without interfering with those sounds.

(3) *Noisy speech—Case 3.* There were difficulties in separating the weak ending /s/, such as in “cats” and “let’s, from the background noise.” Whenever this case occurred, the Best Basis Algorithm produced a 32 ms time window with the first two largest coefficients at frequencies less than 125 Hz. Although the phoneme “/s/” is located at frequencies higher than 125 Hz, its energy was too small to be differentiated from that of the noise. Thus, the data contained in the phoneme /s/ is identified as noise only and disregarded before the compression step. Figure 6.12 illustrates this case.



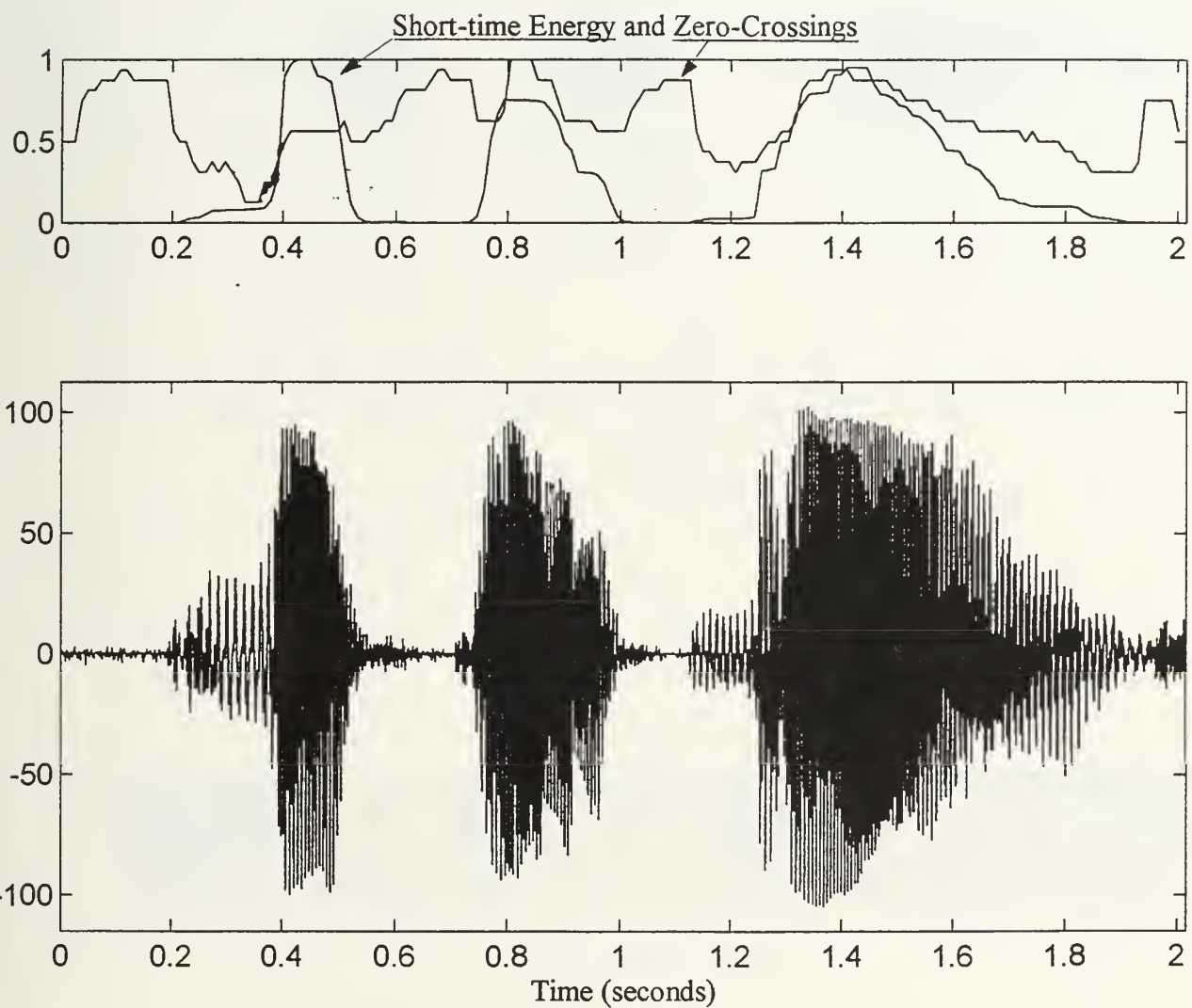


Figure 6.2 Sentence "This Place Blows," male native speaker; top plot: Short-time energy, zero-crossing representation; bottom plot: Time domain representation,  $f_s = 8$  KHz



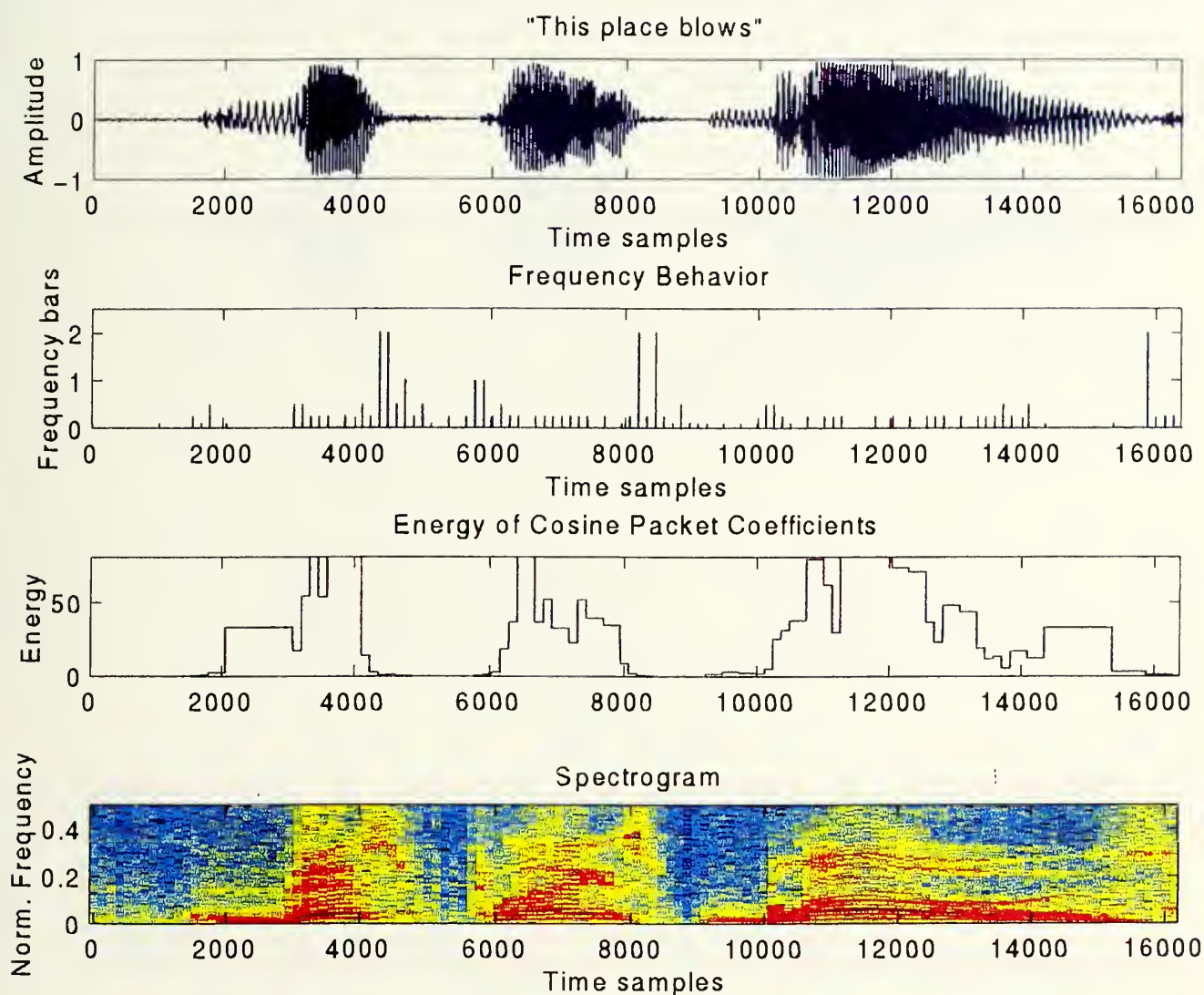


Figure 6.3 Sentence "This Place Blows," male native speaker, "compcp" implementation;  
 (a) Time domain plot; (b) Frequency behavior plot according to Table 6.1; (c) Energy  
 plot; (d) Spectrogram, using a Hanning time window of length 256 samples and  
 overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz



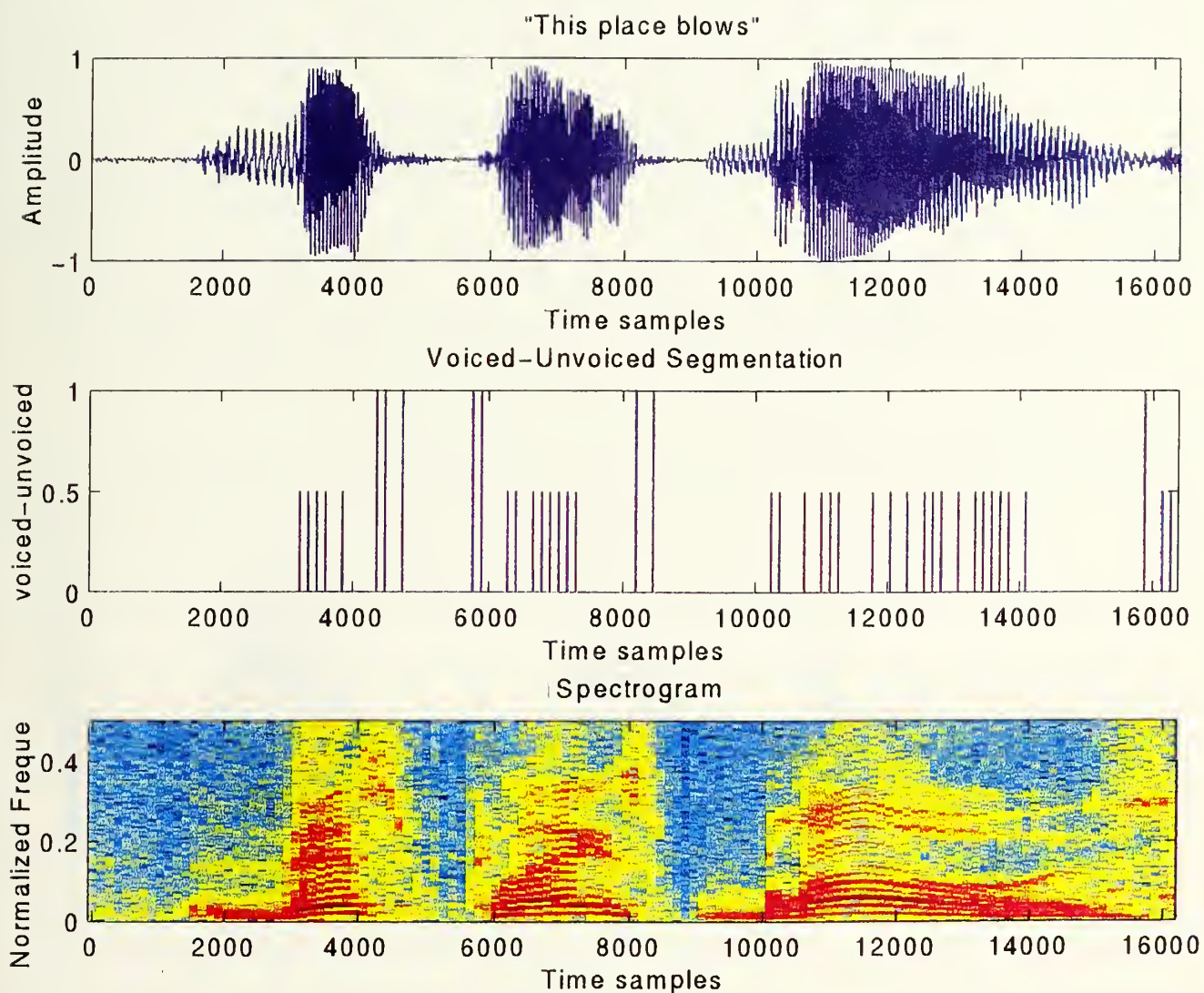


Figure 6.4 Sentence “This Place Blows,” male native speaker, “compcp” implementation; (a) Time domain plot; (b) Voiced-unvoiced segmentation; (c) Spectrogram, using a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz





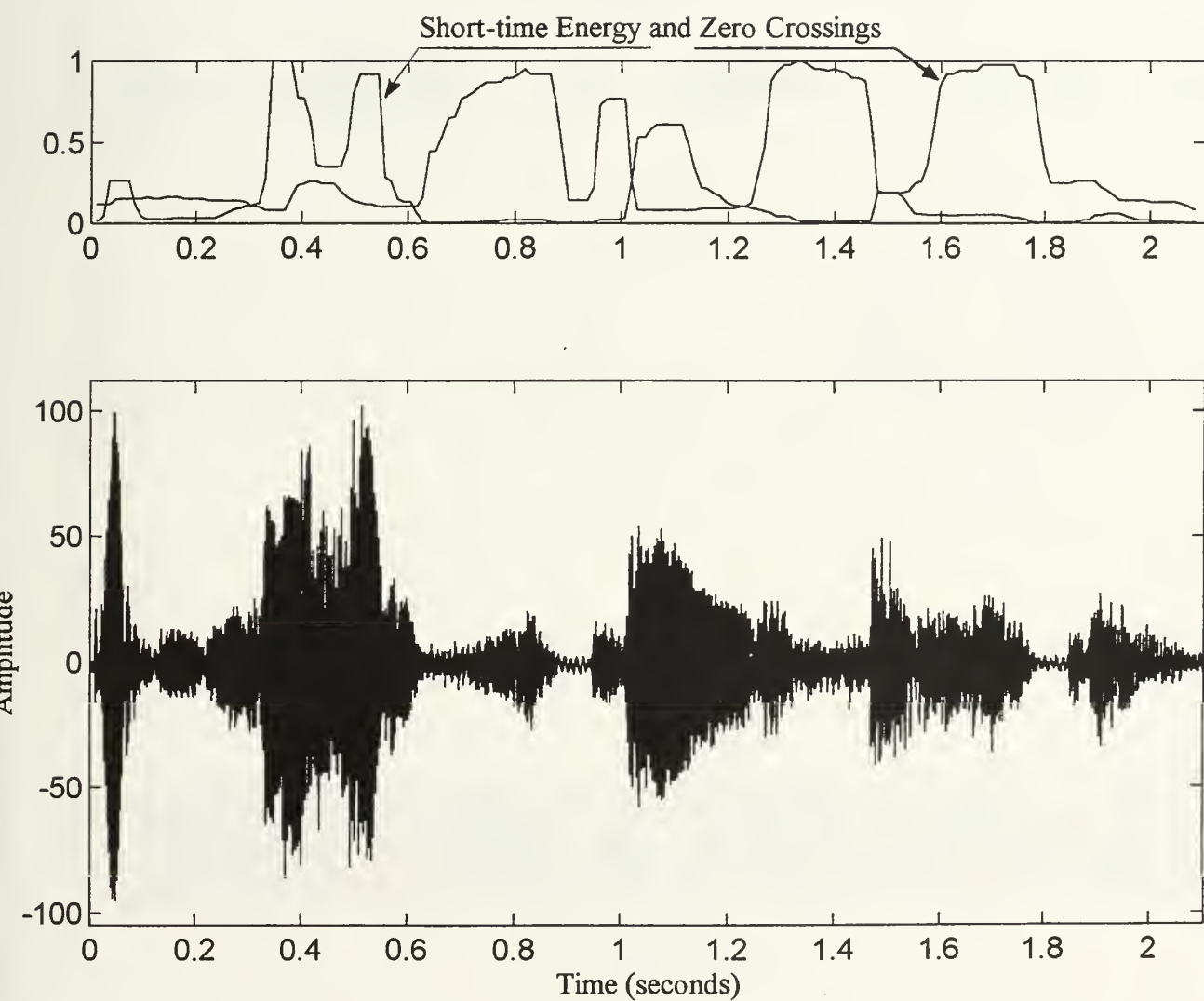


Figure 6.5 Sentence "Be Nice to Your Sister," female native speaker; Top plot: Short-time energy and zero-crossing; bottom plot: Time domain plot





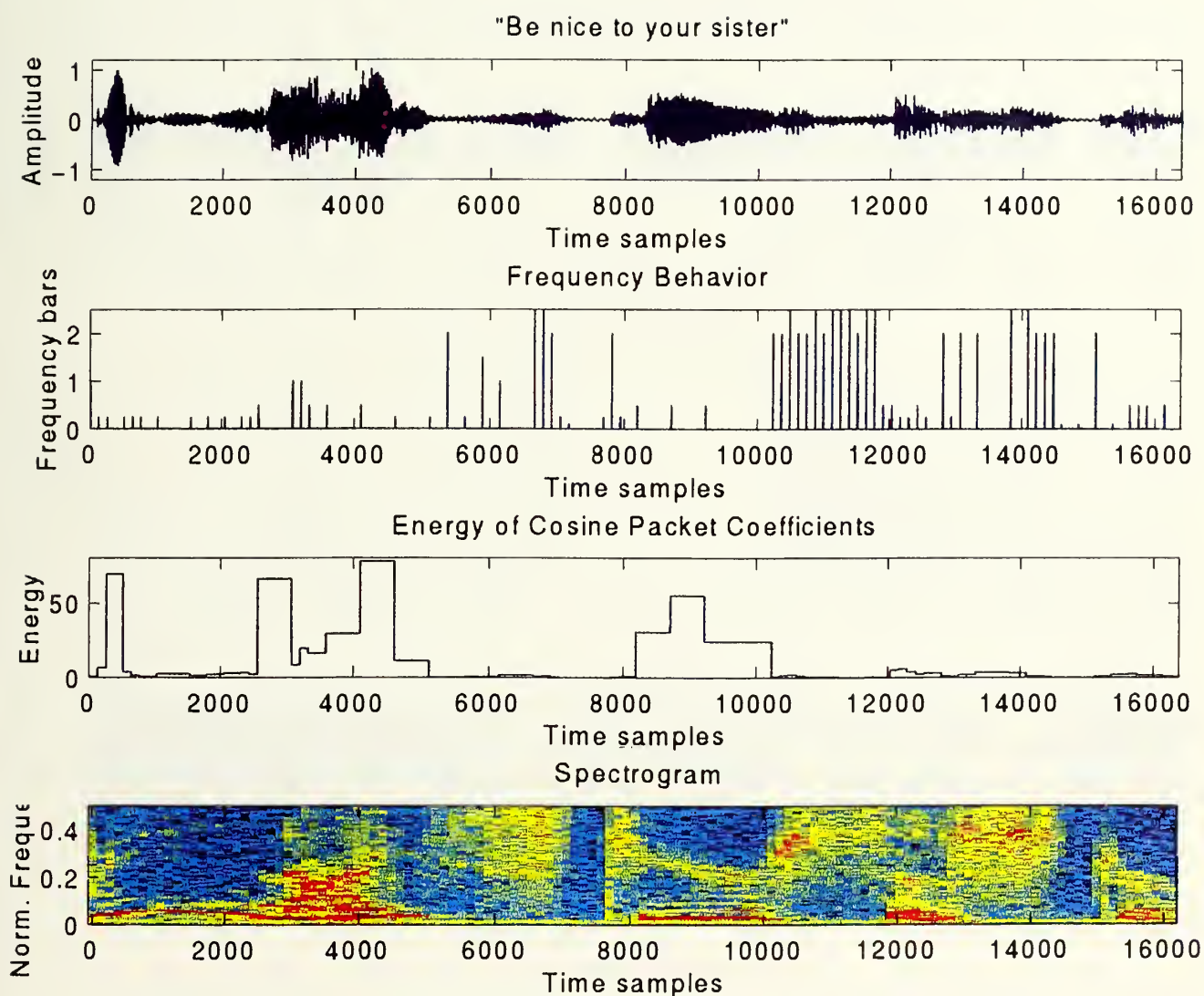


Figure 6.6 Sentence "Be Nice to Your Sister," female native speaker, "compcp" implementation; (a) Time domain plot; (b) Frequency behavior plot, according to Table 6.1; (c) Energy plot; (d) Spectrogram, using a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz



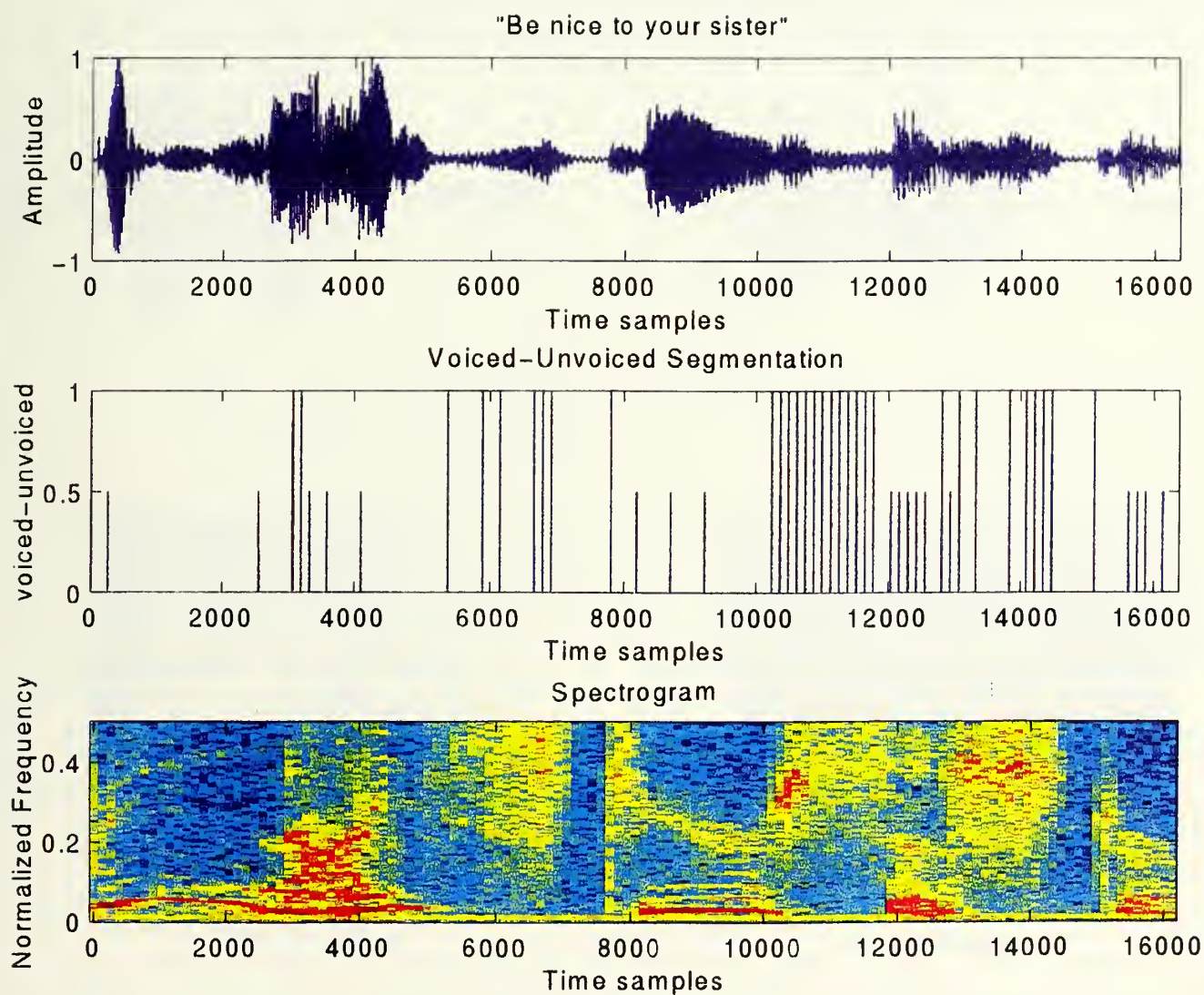


Figure 6.7 Sentence "Be Nice to Your Sister," female native speaker, "compcp" implementation; (a) Time domain plot; (b) Voiced-unvoiced segmentation; (c) Spectrogram, using a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8\text{KHz}$





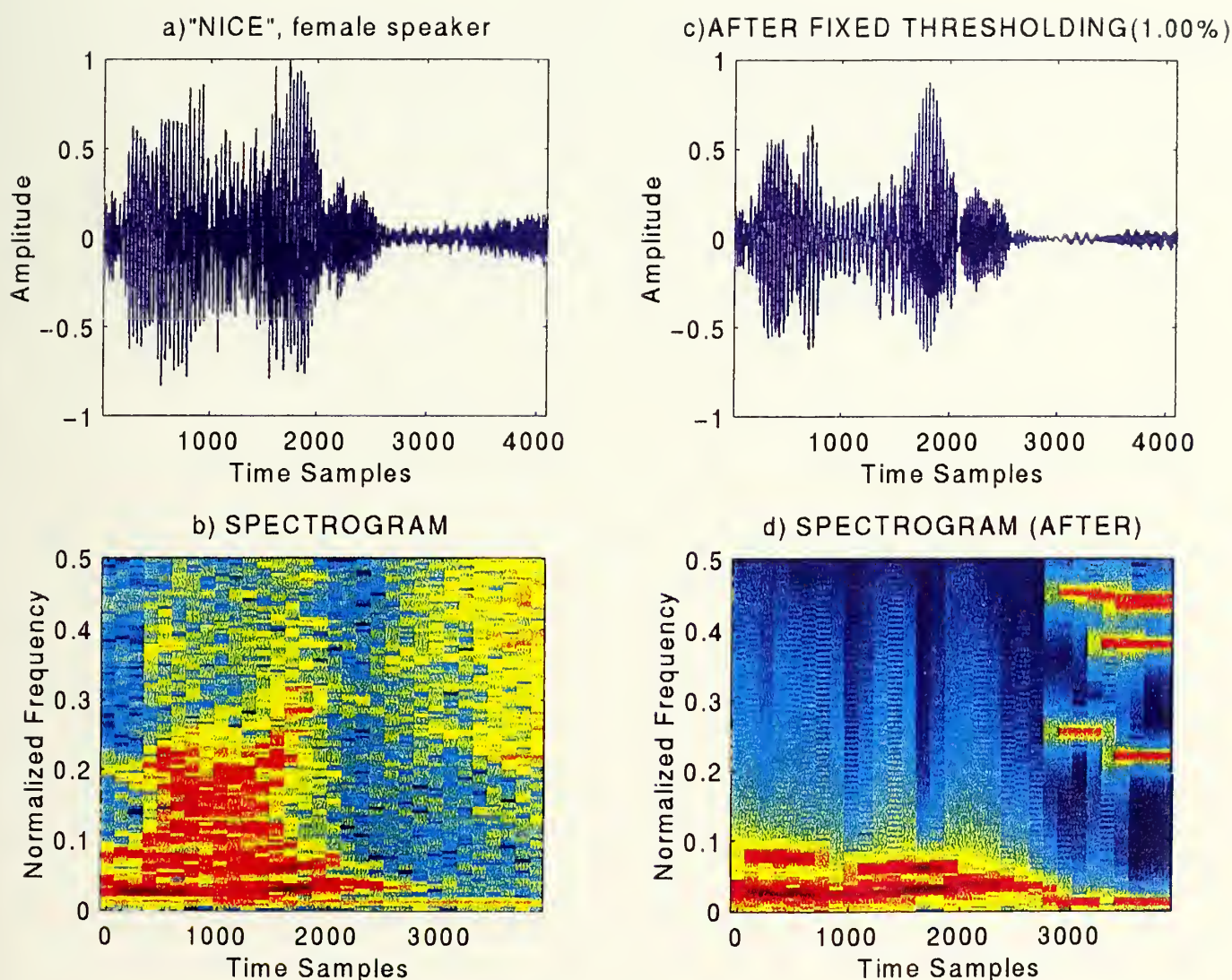


Figure 6.8 Word "Nice," female native speaker, "ndencomp" implementation, fixed thresholding with 1% coefficients kept after compression; (a) Original time plot; (b) Spectrogram of original time speech signal; (c) Plot after fixed thresholding selection of coefficients is applied; (d) Spectrogram of processed signal.(both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8\text{KHz}$ )





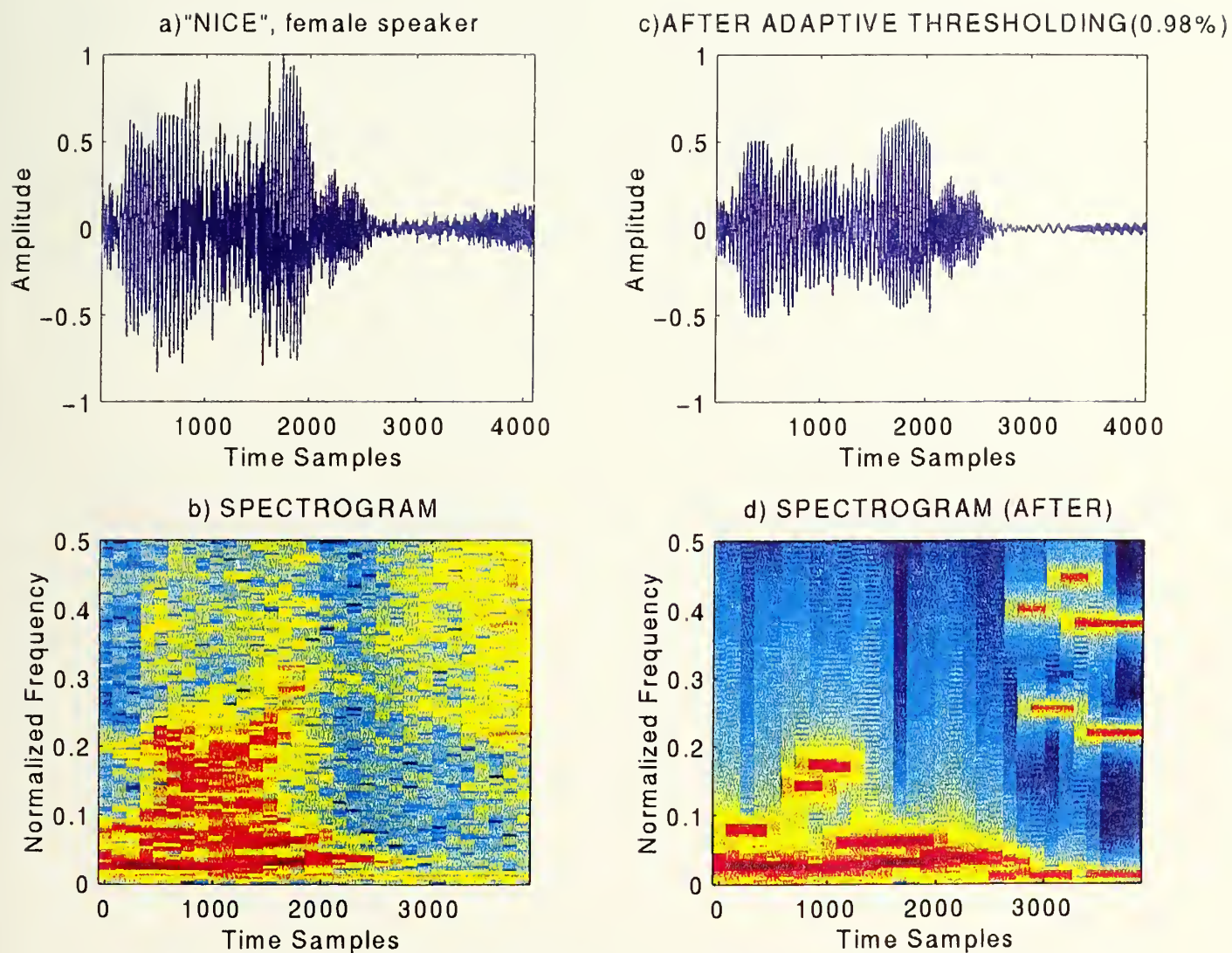


Figure 6.9 Word "Nice," female native speaker, "ndencomp" implementation, adaptive thresholding, with an average of 0.98% CP coefficients kept for compression; (a) Original time domain plot; (b) Spectrogram of original speech signal; (c) Time domain plot of processed signal; (d) Spectrogram of processed signal.(both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 Samples between adjacent windows,  $f_s = 8$  KHz)



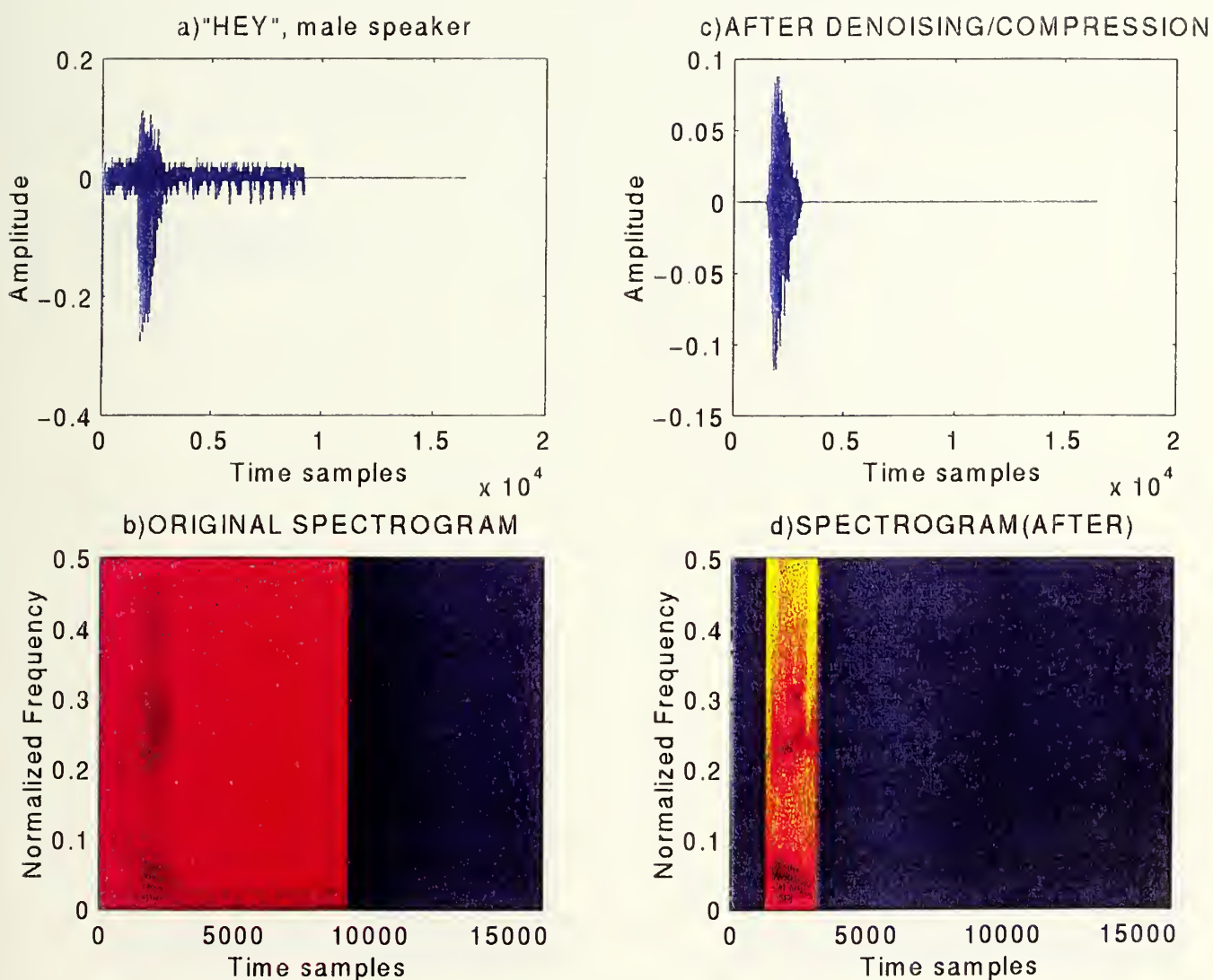


Figure 6.10 Word "Hey," male non-native speaker, "ndencomp" implementation (/h/ lost after denoising scheme when it is identified as noise only); (a) Original time domain plot; (b) Spectrogram of original signal; (c) Time plot after denoising/compression scheme; (d) Spectrogram after denoising/compression scheme. (both spectrograms use a Hanning time window of length 256 Samples and overlapping of 128 samples between adjacent windows,  $f_s = 8\text{KHz}$ )



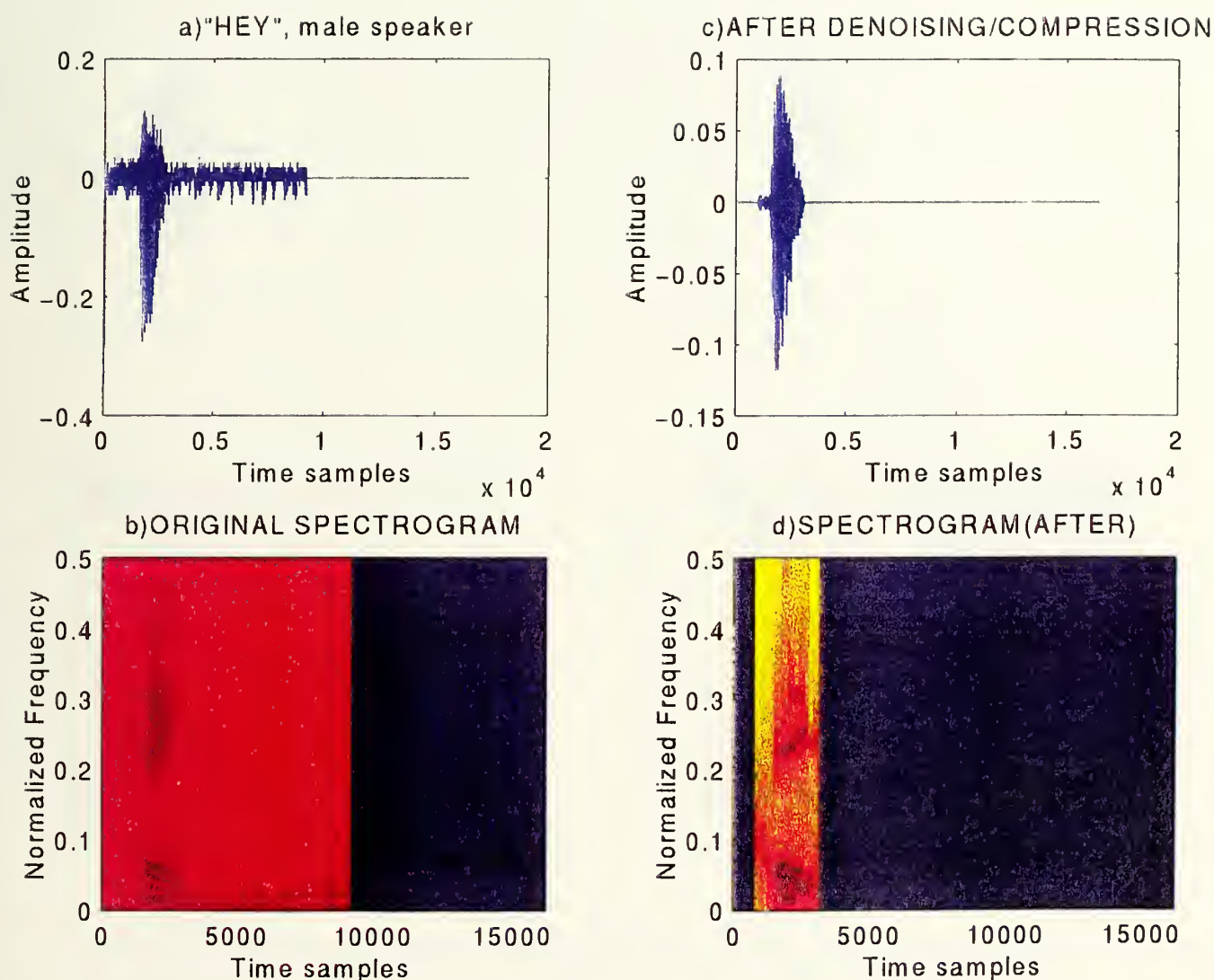


Figure 6.11 Word "Hey," male non-native speaker, "ndencomp" implementation (/h/ recovered after denoising scheme when it is identified as a noisy speech); (a) Original time domain plot; (b) Spectrogram after denoising/compression scheme; (c) Time plot after denoising/compression scheme; (both spectrograms use a Hanning time window of length 256 and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)





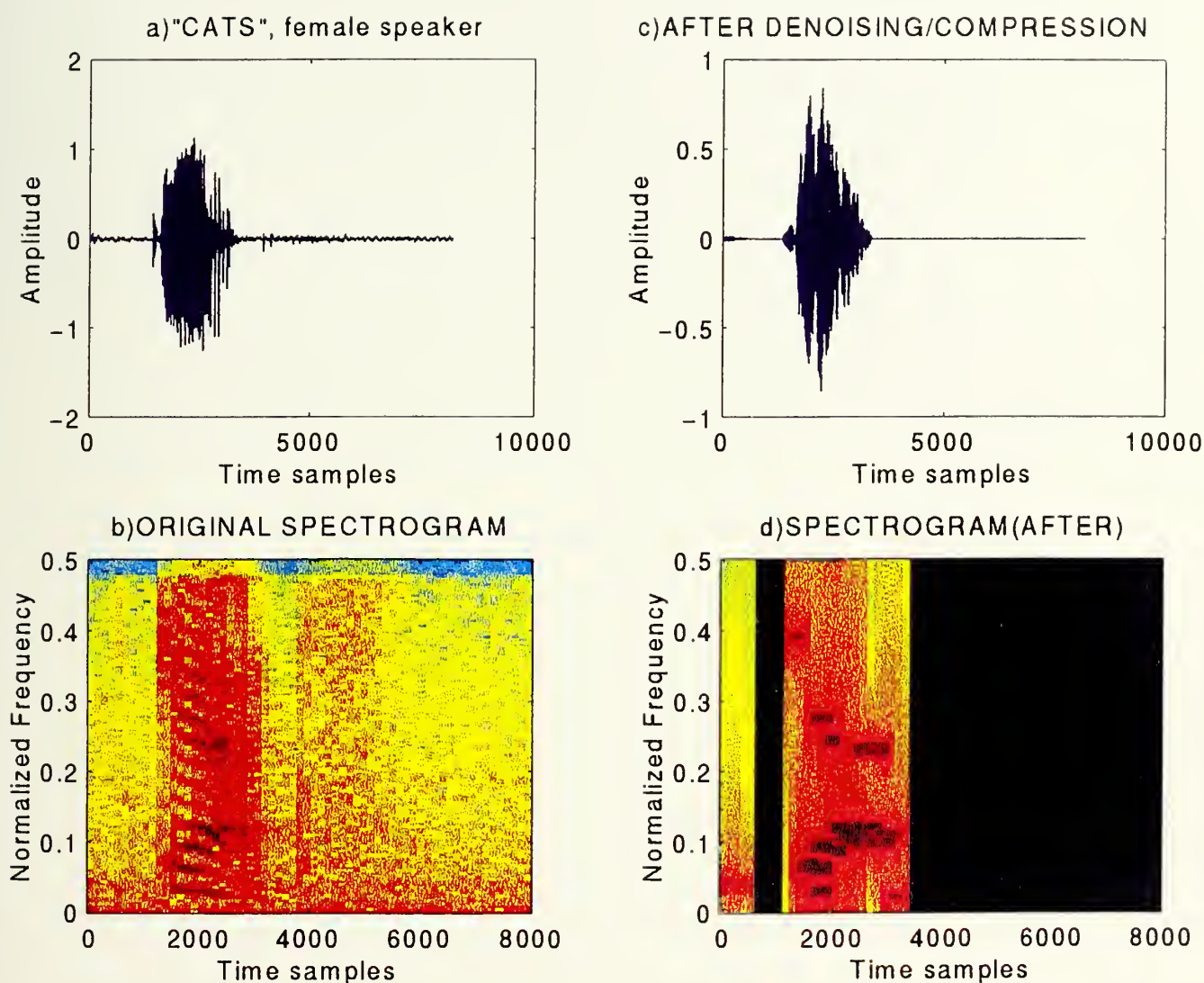


Figure 6.12 Word “Cats,” female non-native speaker, “ndencomp” implementation (/s/ lost after denoising scheme when it is identified as noise only); (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising / compression; (d) Spectrogram after denoising/compression (both spectrograms use a Hanning time window of length 256 and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)









## VII. ENCODING SCHEMES

This chapter is divided into three main sections. The first proposes a quantization scheme to transmit the CP coefficients. The second proposes encoding schemes to transmit the side information, i.e., both the locations and the initial indexes of each segment. The third section presents the coding scheme used to transmit the coefficients vector, the locations vector and the vector containing the initial locations of each segment.

### A. THE QUANTIZATION SCHEME

Once data is available for transmission, the user must quantize and code it. After the compression scheme proves to be efficient, and allows a good quality reconstruction, consideration is given to finding a uniform quantizer that can reproduce efficiently the coefficients to be transmitted [14].

Three different vectors must be sent for speech compression. The first vector contains the cosine packet coefficients. The second contains the location of the coefficients. The third vector contains the initial time locations of each segment. To transmit the first vector, i.e., the coefficients vector, the following is done:

- (1) The data are normalized by dividing all the vectors by the maximum absolute value of all the coefficients. This value turns out to be the scaling factor;

- (2) The whole vector is multiplied by  $QL/2$  (where  $QL$  is the number of quantizing levels selected by the user), and rounded to the closest integer.

By performing these steps, a  $QL$ -level quantizer is built. It has  $QL$  levels due to the normalization and further multiplication by  $QL/2$ , which assures that the positive and negative parts of speech will be always between  $-QL/2$  and  $+QL/2$ .

- (3) The scaling factor, equal to maximum absolute value of all the coefficients, is sent. In the receiver the following steps are to be performed:

- (a) Upon receiving the vector, divide it by  $QL/2$ , recovering the rounded normalized coefficients vector;

(b) Use the scaling factor to recover the amplitudes of the original coefficients. Even without sending the scaling factor, it was possible to recover the coefficients and thus reconstruct the data. The only difference is that the data was scaled in amplitude by a constant factor.

## **B. PROPOSED ENCODING SCHEMES**

### **1. Cosine packet coefficient locations**

To transmit the second vector, i.e., the locations vector, the user first must find the least cost means of transmission. The following example has the sequence of a typical location vector  $L$ :

$$L = [ 1806 \ 1807 \ 1841 \ 1842 \ 1847 \ 1930 \ 1934 \ 1935 \ 2020 \ 2021 \ 2062 \ 2147 \ 2148 \ 218 \ 2192 \\ 2193 \ 2274 \ 2318 \ 2320 \ 2322 \ 2328 \ 2406 \ 2413 \ 2414 \ 2510 \dots ] .$$

Note that there are small differences between some values in this sequence, while larger jumps take place less often. This is because the small differences occur within the same segment, and the larger differences indicate a change from one segment to the adjacent one. Thus, the differences between successive locations are encoded, since they require a smaller number of bits. The differential locations vector correspondent to the locations vector above is given by the vector  $DL$  below:

$$DL = [ 1806 \ 1 \ 34 \ 1 \ 5 \ 83 \ 4 \ 1 \ 85 \ 1 \ 41 \ 85 \ 1 \ 40 \ 4 \ 1 \ 81 \ 44 \ 2 \ 2 \ 6 \ 78 \ 7 \ 1 \ 96 \dots ] .$$

As a result of sending the differences, it is also necessary to send the value for the first location, to allow for an exact reconstruction of the coefficients locations during the decoding process.

## 2. Segment Indexes

The third vector to transmit is the vector that corresponds to the indexes of each segment. The Best Basis Algorithm selects the basis by searching for the minimum entropy representation. When the length of each new window is obtained, the algorithm outputs the two parameters “b” and “d,” which allow the beginning index of the next time window to be computed. The expression for obtaining index “i” is as follows:

$$i = \frac{b}{2^d} \cdot n + 1, \quad (7.1)$$

where  $n$  is the original length of each window. Since the parameters “b” and “d” are small numbers, composed of one or two digits and, therefore, much smaller than the indexes themselves, it is a good idea to transmit the parameters instead of the indexes. Thus, the two vectors “nde” and “nbe,” which are composed of the parameters “b” and “d” of each time window, are transmitted. For example, suppose the vectors nde and nbe are given as follows:

$$\text{nde} = [ 4 \ 6 \ 6 \ 5 \ 5 \ 6 \ 6 \ 5 \ 6 \ 6 \ 5 \ 5 \ 4 \ 4 \ 2 \ 3 \ 6 \ 6 \ 5 ] .$$

$$\text{nbe} = [ 0 \ 4 \ 5 \ 3 \ 4 \ 10 \ 11 \ 6 \ 14 \ 15 \ 9 \ 10 \ 11 \ 6 \ 7 \ 2 \ 6 \ 56 \ 57 ] .$$

Considering  $n = 8192$  time samples, the corresponding vector I containing the initial locations of the first eight segments is given by :

$$I = [ 1 \ 512 \ 640 \ 768 \ 1024 \ 1280 \ 1408 \ 1536 \dots ] .$$

To reconstruct the locations vector of the non-zero coefficients, the receiver works on the received vector of differential locations DL and reconstructs L. The reconstructed vector is then called RL.

Once the locations of non zero coefficients (vector RL) are available, along with the locations of the beginning of each new segment (vector I), the receiver will be able to apply the DCT transform to reconstruct the speech signal.

### C. CODING SCHEMES

After quantization, the coefficients vector is encoded using Huffman Coding [14], which minimizes the total number of bits by assigning more bits to less frequent symbols and less bits to more frequent ones. The vectors  $n_{de}$  and  $n_{be}$  are transformed into only one vector and passed through the Huffman Coder. The inputs include the number of symbols and the probabilities of each one, whereas the outputs from the Huffman Coder are the coded words and average length of the symbols. In order to perform the quantization step and also compute the probabilities of occurrences of each symbol to be coded, the function `quantx.m`, given in the Appendix, was implemented. That function receives the original vector, the number of levels desired for quantization, and returns the quantized vector and the probabilities in descending order, as required by the Huffman Coder (the Huffman Coder used is given in the Appendix). The code was adapted as a function to be called whenever this step is necessary. Finally, the exact number of bits necessary to encode the differential locations vector (DL) is computed.



## **VIII. TESTS AND RESULTS**

### **A. INTRODUCTION**

This chapter describes the procedures that are used to test the compression and encoding schemes. First, the basic compression scheme results are presented. Next, the combined denoising/compression schemes are given. Then, encoding performances, which are used to transmit the compressed information, are presented. Finally, the Cosine Packet compression scheme performances are compared with those obtained using the the related Wavelet Packet Transform.

### **B. COMPRESSION SCHEME RESULTS**

The compression-only scheme is first applied to “clean” speech to evaluate its performance. To test this scheme on isolated words, we use the words “project,” “cataratas,” and the segment “encyclope,” extracted from the word encyclopedia. This compression scheme is also implemented in the following two sentences:

“ Be nice to your sister,” spoken by a female native speaker; and

“ This place blows,” spoken by a male native speaker.

#### **1. Description**

The testing software requires the user to input the following:

(1) The gender of the speaker. This information is required since the pitch for a female speaker occurs at a higher frequency than that of a male speaker;

(2) Word or sentence to be compressed;

(3) Maximum depth used for the cosine packet time splitting, which in turns fixes the minimum size of the window;

The following outputs are provided:

(1) The mean square error between the original and the reconstructed speech signal;

(2) The number of non-zero cosine packet coefficients in the original signal (ONCOEF);

(3) The number of non-zero cosine packet coefficients selected by the compression scheme (FNCOEF);

(4) The reconstructed speech signal obtained after compression.

Two different compression implementations were considered, which differ in the number of cosine packet coefficients kept to compress the speech signal. The first compression scheme (implemented in `compcp.m`, given in the Appendix) selects the cosine packet coefficients as follows:

(1) Keep the top 0.5% non-zero coefficients (rounded to the closest integer) in each time window when the speech segment is detected as unvoiced; this percentage means selecting one coefficient out of every interval containing 128 coefficients, one coefficient out of every interval containing 256 coefficients, and so on, according to the result of the rounding process;

(2) Keep the top 1.3% non-zero coefficients (rounded to the closest integer) for each time window of minimum length (16 ms) when the speech segment is identified as voiced; this means selecting two coefficients out of every 128 coefficients, three coefficients out of every interval containing 256 coefficients, and so on;

(3) Keep the top 2.34% non-zero coefficients (rounded to the closest integer) for each time window larger than 16 ms when the speech segment is identified as voiced; this means selecting three coefficients out of every interval containing 128 coefficients, six coefficients out of every interval containing 256 coefficients, and so on.

The second compression scheme (implemented in `necompcp.m` and given in the Appendix) uses the following schemes to compress the speech signal:

(1) Keep the top 0.5% non-zero coefficients (rounded to the closest integer) in each time window when the speech segment is unvoiced; this means selecting one coefficient out of every interval containing 128 coefficients, one coefficient out of every interval containing 256 coefficients, and so on;

(2) Keep the top 1.3% non-zero coefficients (rounded to the closest integer) for each time window when the speech segment is identified as voiced; this means selecting two coefficients out of every interval containing 128 coefficients, three coefficients out of every interval containing 256 coefficients, and so on.

## 2. Experimental Results

Results obtained for the two compression schemes are presented in Table 8.2. In Chapter VI, Figures 6.8 and 6.9 present time domain plots and spectrograms for the Adaptive Thresholding compression scheme considered in this section. The parameters used to measure degradation due to the compression scheme are:

- (1) The mean square error (MSE) between the original and the reconstructed speech signal;
- (2) A subjective evaluation made by five different users of the quality of the reconstructed signal when compared to the quality of the original signal. The evaluation was graded on a scale from 1 to 5, according to Table 8.1.

GRADE	Speech Quality	Level of Distortion
5	Excellent	Imperceptible
4	Good	Just perceptible but not annoying
3	Fair	Perceptible and slightly annoying
2	Poor	Annoying but not objectionable
1	Unsatisfactory	Very annoying and objectionable

Table 8.1 Mean opinion score table

- (3) The ratio between the number of non-zero cosine-packet coefficients kept after compression and the total number of initial non-zero coefficients obtained with the cosine packet decomposition (ONCOEF/FCOEF%).

### 3. Comments

Note the slightly higher speech quality mean grades assigned to code compcp.m, which also presents a slightly higher percentage of coefficients kept (i.e., a lower compression ratio). Experiments showed that fixed thresholding selects 1% of the set of coefficients, and leads to the distortion of voiced phonemes (e.g., /i/ ends up sounding like /u/ in the word “nice”). The “after compression” spectrogram included in the bottom right of Figure 6.8 showed that the higher formant section of the phoneme /i/ has not been preserved in the compression. By comparison, Figure 6.9 showed the results obtained using an adaptive thresholding scheme, which selects more coefficients for the voiced segments while keeping a smaller total percentage of coefficients (0.98%). The after-compression spectrogram shown in Figure 8.2 shows that the high formants of the phoneme /i/ are better preserved, leading to a better reconstruction of the voiced phoneme.

<i>Parameters</i>	<i>“Project”</i>	<i>“Cataratas”</i>	<i>“Encyclope”</i>	<i>“Issos”</i>	<i>“Assos”</i>	<i>“Be Nice”</i>	<i>“This Place”</i>
<b>Code: NECOMP.M</b>							
MSE	0.0045	0.0315	0.0127	0.0074	0.0136	0.0070	0.0432
ONCOEF	8192	8192	8192	8192	8192	16384	16384
FNCOEF	124	100	104	100	100	210	216
% ONCOEF/ FNCOEF.	1.51	1.22	1.27	1.22	1.22	1.28	1.32
Speech quality mean grade	2.2	2.4	2.8	2.8	2.2	3.2	3.0
<b>Code: COMPCP.M</b>							
MSE	0.0038	0.0313	0.0121	0.0057	0.0115	0.005 2	0.029 7
ONCOEF (original # of coeff.>0)	8192	8192	8192	8192	8192	16384	16384
FNCOEF (final # of coeff. >0)	181	109	126	129	115	139	228
% ONCOEF/ FNCOEF	2.21	1.33	1.54	1.57	1.40	0.85	1.39
Speech quality mean grade	2.6	2.6	2.8	3.0	2.2	3.4	3.2

Table 8.2 Compression only results



## C. DENOISING-COMPRESSION RESULTS

### 1. Description

Next, we consider the application of a combined denoising and compression scheme designed to minimize the effects of narrowband equipment noise in a few isolated words and sentences. The isolated words used in these tests are:

- “Be”, spoken by a female and by a male speaker;
- “Cats”, spoken by a female speaker;
- “Hey”, spoken by a female and by a male speaker;
- “Met”, spoken by a female speaker; and
- “Pay”, spoken by a female speaker.

The sentences used are:

- “Hello, my name is Roberto, today is Tuesday;” and
- “Bye, guys, I’m going back to Brazil”, both spoken by a male speaker.

Two different implementations for the denoising scheme are considered: The first is implemented in `ndencomp.m` and the second is implemented in `encp6.m` (both are listed in the Appendix). The noise identification and denoising process for each implementation can be found in Chapter VI, Section E, and in the Appendix. Details regarding the compression scheme for both implementations can be found in the Appendix. Table 8.3 presents the compression results for tests applied on the same “clean” words of the previous section, but using the codes `ndencomp.m` and `encp6.m`.

### 2. Results

The parameters used to evaluate the denoising/compression scheme are identical to those defined for the compression-only scheme, with the exception of the mean square error (MSE). This parameter was omitted because the denoising step produced a greater difference between the original and the reconstructed signals. The performance results are presented in Table 8.4.

<i>Parameters</i>	<i>“Project”</i>	<i>“Cataratas”</i>	<i>“Encyclope”</i>	<i>“Issos”</i>	<i>“Assos”</i>	<i>“Be Nice”</i>	<i>“This Place”</i>
<b>Code: NDENCOMP.M</b>							
ONCOEF	8192	8192	8192	8192	8192	16384	16384
FNCOEF	189	153	152	129	145	277	224
% ONCOEF/ FNCOEF	2.31	1.87	1.86	1.87	1.77	1.69	1.37
Speech quality mean grade	2.5	3.0	3.0	3.2	2.6	3.3	3.2
<b>Code: ENCP6.M</b>							
ONCOEF	8192	8192	8192	8192	8192	16384	16384
FNCOEF	188	149	152	129	143	270	264
% ONCOEF/ FNCOEF	2.29	1.82	1.86	1.86	1.75	1.65	1.61
Speech quality mean grade	2.5	3.3	3.2	3.4	2.8	3.5	3.4

Table 8.3 Compression results utilizing codes ndencomp.m and encp6.m

The speech quality mean grade was computed for the following speech data: The words “Be,” female speaker, and “pay,” male speaker, and the sentences “Hello, my name is Roberto ...” and “Bye, guys, I’m going back ...” These results are presented in Table 8.5.



<i>Parameters</i>	<i>"Be," Female</i>	<i>"Be," Male</i>	<i>"Cats," Female</i>	<i>"Hey," Female</i>	<i>"Hey," Male</i>	<i>"Met," Female</i>	<i>"Pay," Female</i>	<i>"Pay," Male</i>	<i>"Hello , My Name"</i>	<i>"Bye..."</i>
<b>Code: NDENCOMP.M</b>										
ONCOEF	6270	5248	8190	8188	9342	8190	7296	7294	32768	24574
FNCOEF	59	41	143	64	42	37	49	46	569	318
% ONCOEF/ FNCOEF	0.94	0.78	1.75	0.78	0.44	0.45	0.67	0.63	1.74	1.29
<b>Code: ENCP6.M</b>										
ONCOEF	6272	5248	8192	8192	9344	8192	7296	7296	32768	24576
FNCOEF	72	48	27	65	35	40	55	34	573	348
% picked	1.15	0.91	0.33	0.79	0.37	0.49	0.75	0.47	1.75	1.42

Table 8.4 Denoising/compression results

<i>SPEECH</i>	<i>ndencomp</i>	<i>encp6</i>
"Be" (female speaker)	2.2	3.4
"Pay" (male speaker)	2.6	2.6
"Hello, my name is ..."	3.2	3.2
"Bye, guys ..."	2.4	2.6

Table 8.5 Speech quality mean grades

### 3. Comments

We note that the overall speech quality mean grades in Table 8.3 are slightly increased when compared to those from Table 8.2. We also note that the ONCOEF/FNCOEF percentages in Table 8.4 are small, since large sections of data are identified as noise-only. Thus they are not retained for compression by the denoising step. Results obtained for both denoising/compression schemes show slightly better speech quality for the encp6.m implementation than for the ndencomp.m implementation.

#### *a. Word “be”*

Both denoising/compression schemes produce good results for the word “be” for male and female speakers. The plots in Figure 8.1 show the efficiency of the algorithm in both the time and frequency domain. The quality of the reconstructed speech is good, as illustrated by the grades assigned by five native listeners.

#### *b. Word “Hey”*

For male and female cases, both denoising/compression schemes produce good results (Figures 8.2 and 8.3). The quality of the reconstructed speech is high, as confirmed by the listening tests. Note that the /h/ sound in the female speech has a higher frequency than that of the male voice. The denoising schemes also allow the phoneme /h/ to be differentiated from the noisy background environment.

#### *d. Word “met”*

Both denoising/compression schemes produce a good reconstruction of “/me/” and a poor reconstruction of the phoneme /t/, which is reconstructed sounding like a “/d/.” This degradation is due to the combination of too few coefficients kept for compression in this section of the word and a noisy background.

*e. Word “Pay”*

The reconstructed sound quality is better for the male case (Figures 8.4 and 8.5). Again, this was due to too few coefficients being kept. The sound /p/ has its first two largest CP coefficients below 400 Hz and around 1,000 Hz, respectively. Higher energy is concentrated in the lower frequency coefficients. When spoken by a female, the higher frequency coefficients get less energy compared to others not so important from the lower frequency portion. For that reason, fewer coefficients from the higher frequency portion are kept, leading to a poorer sound than the male version.

*f. Sentence “Hello, my name is Roberto, today is Tuesday”*

The spectrograms in Figure 8.6 show that the main signal energy is preserved, and that denoising occurs in the correct time intervals.

*g. Sentence “Bye, guys, I’m going back to Brazil ”*

For this sentence, both denoising-plus-compression schemes result in a good reconstruction. It is possible to observe in the spectrograms of Figure 8.7 that the algorithm picks up the important cosine packet coefficients. In this case, no significant amount of denoising was done due to the high quality of the original speech. However, it is worth comparing the effects of the denoising schemes. Note that, in using `ndencomp.m` the resultant signal is divided more by noisy intervals than when using `encp6`. In the listening tests for both sentences, the mean grade assigned to the reconstruction using `ndencomp.m` is better than the one assigned when using `encp6.m`. Basically, the unvoiced sounds had a better reconstruction using the former code, whereas the latter code produced some distortion, leading to what was called a mechanical sound by some listeners.

## **D. ENCODING SCHEMES RESULTS**

### **1. Description**

The data used for these tests consist of twelve speech sequences of lengths 8192 (ten words and/or sounds), 32768 (one sentence) and 65536 (one dialogue). The software used for this set of tests includes voiced-unvoiced segmentation, denoising, compression, and encoding steps. Both denoising/compression schemes are used in these tests. The coding software is presented in the Appendix. The minimum window size is 16 milliseconds. The compression ratio between the total number of bits after encoding and the total original number of bits, is used to evaluate the performance of the encoding scheme. The original number of bits is computed by multiplying the number of bits used to represent each incoming sample (the samples had 8 bits and were PCM compressed) by the original number of samples. For example, for each of the ten sequences of length 8192, the original number of bits is  $8192 \cdot 8 = 65,536$  bits per speech sequence. The following speech sequences are used in our tests:

- (a) “BE,” spoken by a female speaker;
- (b) “HEY,” spoken by a female speaker;
- (c) “MET,” spoken by a female speaker;
- (d) “PAY,” spoken by a female speaker;
- (e) “CATS,” spoken by a female speaker;
- (f) Word “PROJECT,” spoken by a male speaker;
- (g) Word “CATARATAS,” spoken by a male speaker;
- (h) Sound or partial word “ENCYCLOPE,” spoken by a male speaker;
- (i) Sound “ASSOS,” spoken by a male speaker;
- (j) Sound “ISSOS,” spoken by a male speaker;
- (k) Sentence “Bye guys, I’m going back to Brazil,” male speaker;
- (l) Dialogue from a telephone conversation, male and female speakers.

## 2. Results

A measure of distortion is obtained by comparing the quality between the original speech signal and the reconstructed signal. (“Speech quality” in Table 8.1).

To evaluate the efficiency of the encoding scheme, the following parameters are chosen:

(1) The  $\overline{\text{COMPRATIO}}$ , defined as one minus the ratio between the total number of bits after compression and the total number of bits in the original signal;

(2) The mean square value of the quantization error.

The performance results for the encoding scheme are presented in the Table 8.6. All results are based on the denoising/compression implementation `ndencomp.m`, except for the words “hey” and “met,” which use `encp6.m`.

SPEECH	SPEECH QUAL	$\overline{\text{COMPRATIO}}$ %	MSE
“Be”	2.6	98.70%	$5.12e^{-7}$
“Hey”	3.2	98.56%	$9.32e^{-6}$
“Met”	3.2	98.85%	$5.93e^{-6}$
“Pay”	3.0	99.17%	$5.22e^{-7}$
“Cats”	3.4	98.59%	$1.01e^{-5}$
“Project”	3.2	97.87%	$1.06e^{-5}$
“Cataratas”	3.4	97.65%	$4.61e^{-5}$
“Encyclope”	3.8	97.64%	$2.86e^{-4}$
“Assos”	3.2	97.87%	$3.62e^{-5}$
“Issos”	4.0	98.05%	$2.16e^{-5}$
“Bye, guys...”	2.4	98.10%	$2.707e^{-5}$
Tel. conversation	2.6	98.06%	$2.843e^{-5}$

Table 8.6 Encoding results, 64-level quantizer.



The mean speech quality grade assigned is 3.2 (see MOS Table 8.1). This means perceptible and slightly annoying. We also note the high values of compression ratios and the very small values of MSE, which corresponds to the mean square quantization error.

The compression ratio is calculated in the following way. Eight bits are used for each original sample of data, since that is the number used to load speech recordings into “Matlab.” The total number of bits is computed by multiplying each average number from the Huffman coder by the corresponding number of samples in the coefficients vector, as well as in the three vectors used to transmit the locations and the window boundaries. The compression ratio is then computed as the ratio between the final total number of bits and the original total number of bits after the encoding process.

Comparing the percentages from this encoding table to the ones from the previous sections (compression and denoising/compression), we note that, although still very low, the numbers from the encoding process are higher ( $\sim 2\%$ ) in comparison to the others ( $\sim 0.85\%$ ). The reason is that, in addition to the cosine packet coefficients, the side information (i.e., the locations of those coefficients) must also be encoded. Thus, even though the number of bits is reduced due to the quantization process, the increase of information to be transmitted makes the number a little higher.

As can be noted from the grades assigned, the encoding process results are good. The only problem are the low-energy coefficients corresponding to unvoiced sounds when submitted to the quantization and rounding processes. Figure 8.8 shows the word “project,” which lost its weak, final /kt/. Even when we change the quantizer to 32 and 64 levels, it is still impossible to recover the final sound.

Figure 8.9 presents the sentence, “Be nice to your sister,” using a 16-level quantizer. We note that the sounds /s/ in “nice”, /t/ in “to,” and /r/ in “your” are lost. However, when the quantizer is changed to 32 levels, the main parts of these sounds are recovered (Figure 8.10). Finally, when the number of levels is increased to 64 (Figure 8.11), the sequence sound is totally reconstructed, and practically no difference is noted between the original and reconstructed sounds.



Similar results are observed for the sentence, “Bye guys, I’m going back to Brazil.” The phoneme /z/ from “guys” is lost with a 16-level quantizer (Figure 8.12), and is recovered with a 32-level quantizer (Figure 8.13). Similarly, when a 16-level quantizer was applied, the sound /h/ in the word “hey” was reconstructed like a /k/, resulting in a word sounding like “kay” (Figure 8.14). By changing to a 32-level quantizer, it was possible to recover the correct sound (Figure 8.15). The sound was even better with a 64-level quantizer (Figure 8.16). Note the sequential progress in the coefficients recovered in Figures 8.14 through 8.16, by comparing the plots (d) and (f).

Two points are worth mentioning. First, after the number of quantizing levels is doubled, the compression ratio does not decrease significantly. For example, for the word “project” (with a higher SNR, an almost “clean” word), when the number of levels is increased from 16 to 32 (i.e. changing from 4 to 5 bits/symbol), the compression percentage changes from 98.36% (1:61.2) to 98.28% (1:58.4). Another example is the word “hey” (also a high SNR). The three compression percentages corresponding to the 16-level, 32-level, and 64-level quantizers are, respectively, 99.23% (1:130.4), 99.16% (1:119.2), and 99.11% (1:113.2), respectively. Thus, a 64-level quantizer is used as a good compromise between quality and compression.

## **E. COMPARISON WITH WAVELET PACKET TRANSFORM**

In this section, the Cosine Packet is compared to the Wavelet Packet-based compression procedure in clean (high SNR) speech. A “clean”( high SNR) speech sequence is chosen, and the results are compared up to only the compression scheme, since the encoding scheme performs basically the same for both cases.

The sentence “Be nice to your sister” is compressed using the Cosine Packet Transform, and the average percentage of non-zero coefficients selected from the original number equals 0.85% for a good reconstruction of the speech. A much poorer

reconstruction results from the Wavelet Packet Transform using the “Daubechies(4)” wavelet basis function. The WT is implemented using the same criteria as those defined for the CPT implementation with the WaveLab Package [17].

The result obtained for this sentence can be analyzed through the time and frequency plots for both schemes given in Figures 8.17 and 8.18. We note that, in the Wavelet Packet Transform, there are “holes” in the time domain. We note also that those “holes” happen to be exactly at the intervals where the energy is lower, i.e., mainly at the unvoiced sounds. This is because the WPT scheme initially splits the signal into given frequency windows. In our example, only the highest 15% coefficients for given frequency ranges are selected during the whole period of time.

By comparison, the CPT splits the signal first in the time domain. Then, for each time frame, a thresholding is applied for the cosine packet coefficients. As a result, although many fewer coefficients are selected, there is no chance of having a time interval not represented. Actually, in this scheme, the holes are in the frequency domain. But, since the transform is good enough to detect the main frequencies contained in each locally stationary portion of the signal, the few cosine packet coefficients preserved at each time interval are sufficient to allow for a good reconstruction of the speech. These results confirm the theoretical expectation of superiority of the CPT over the WPT for speech signal compression applications.



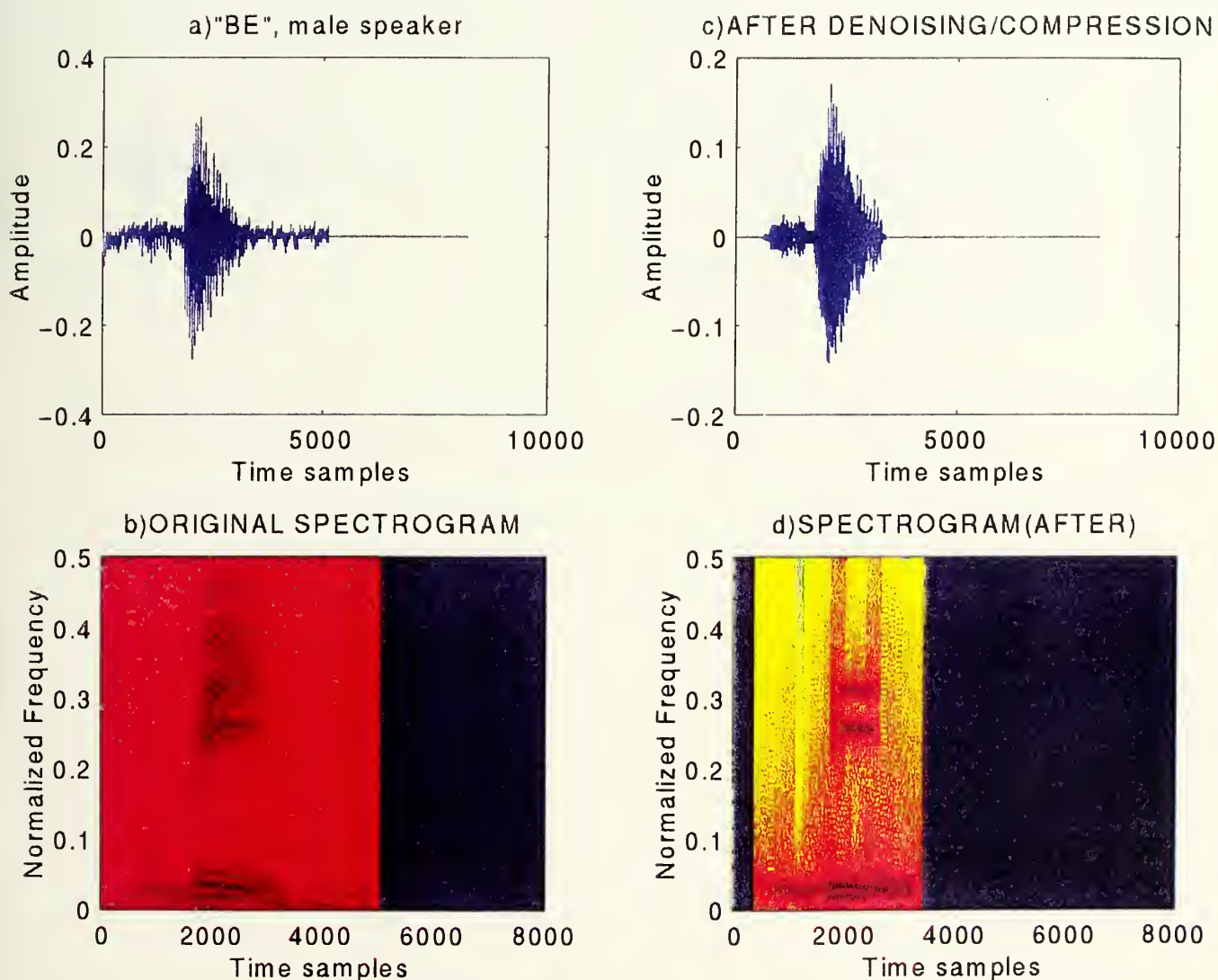


Figure 8.1 Word "Be," male non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original plot; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising/compression (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)



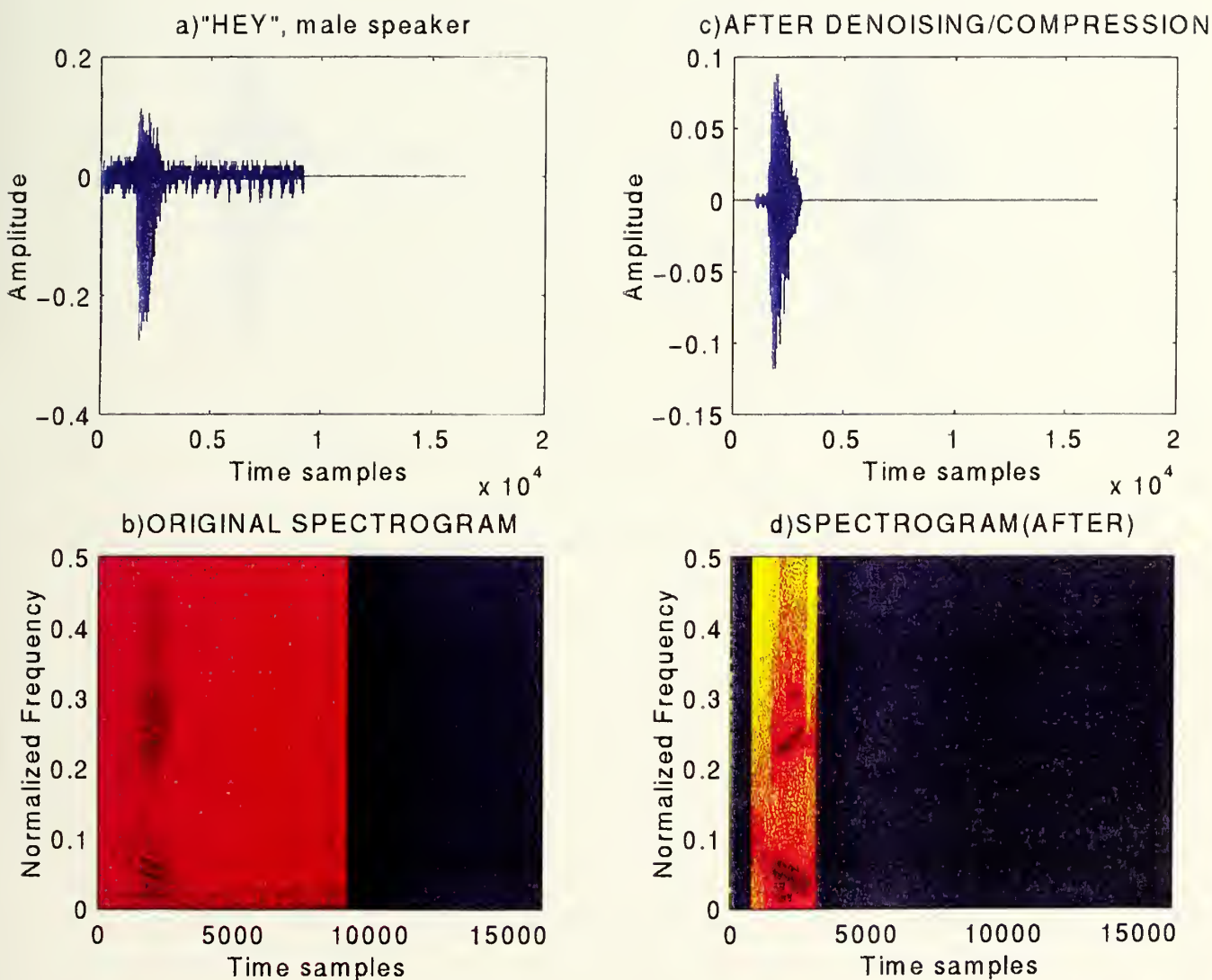


Figure 8.2 Word “Hey,” male non-native speaker, “ndencomp” implementation;  
 (a) Original time domain plot;(b) Spectrogram of original speech; (c) Time domain plot  
 after denoising/compression; (d) Spectrogram after denoising/compression (both  
 spectrograms use a Hanning time window of length 256 samples and overlapping of 128  
 samples between adjacent windows, $f_s = 8$  KHz)





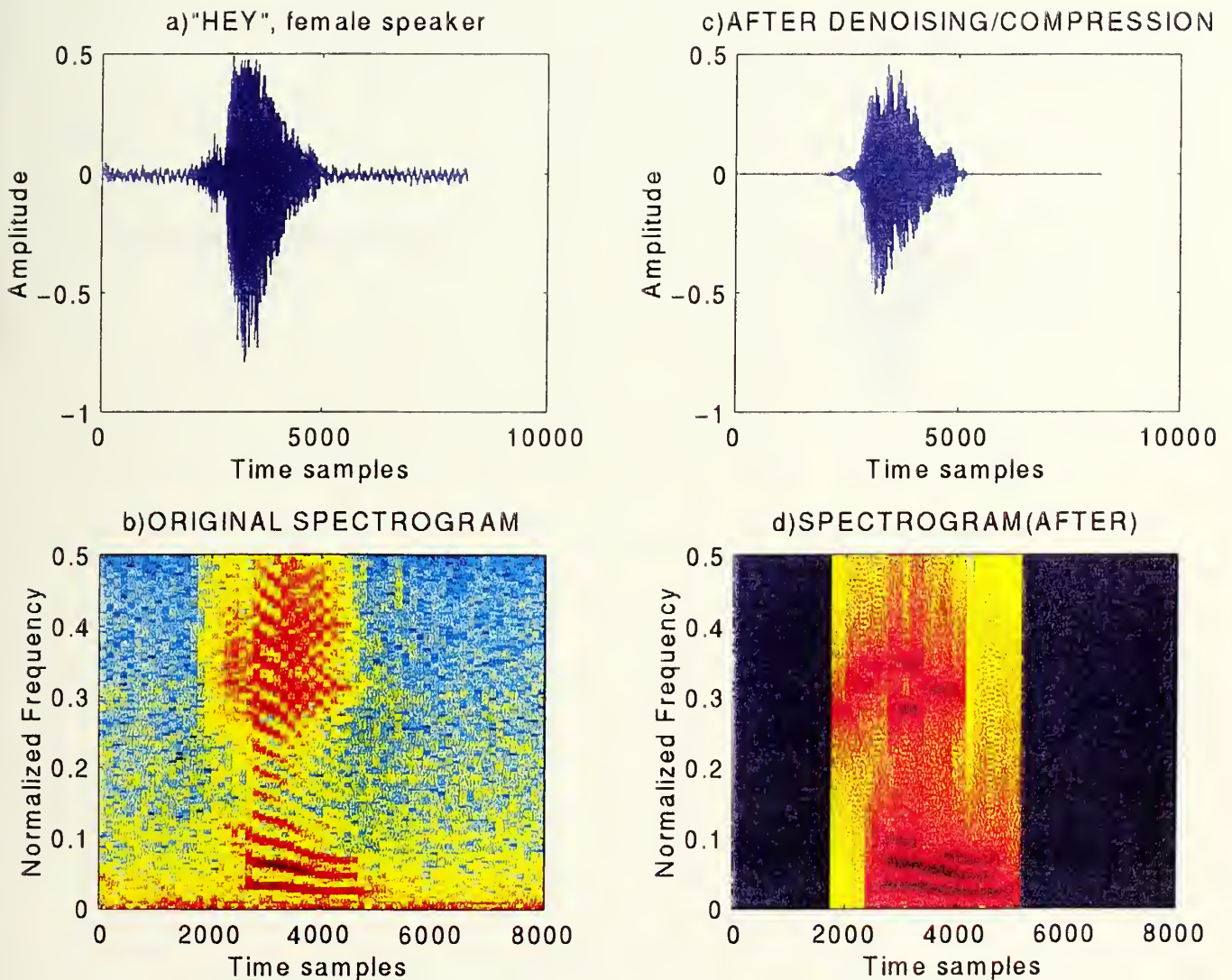


Figure 8.3 Word "Hey," female non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising/compression (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)



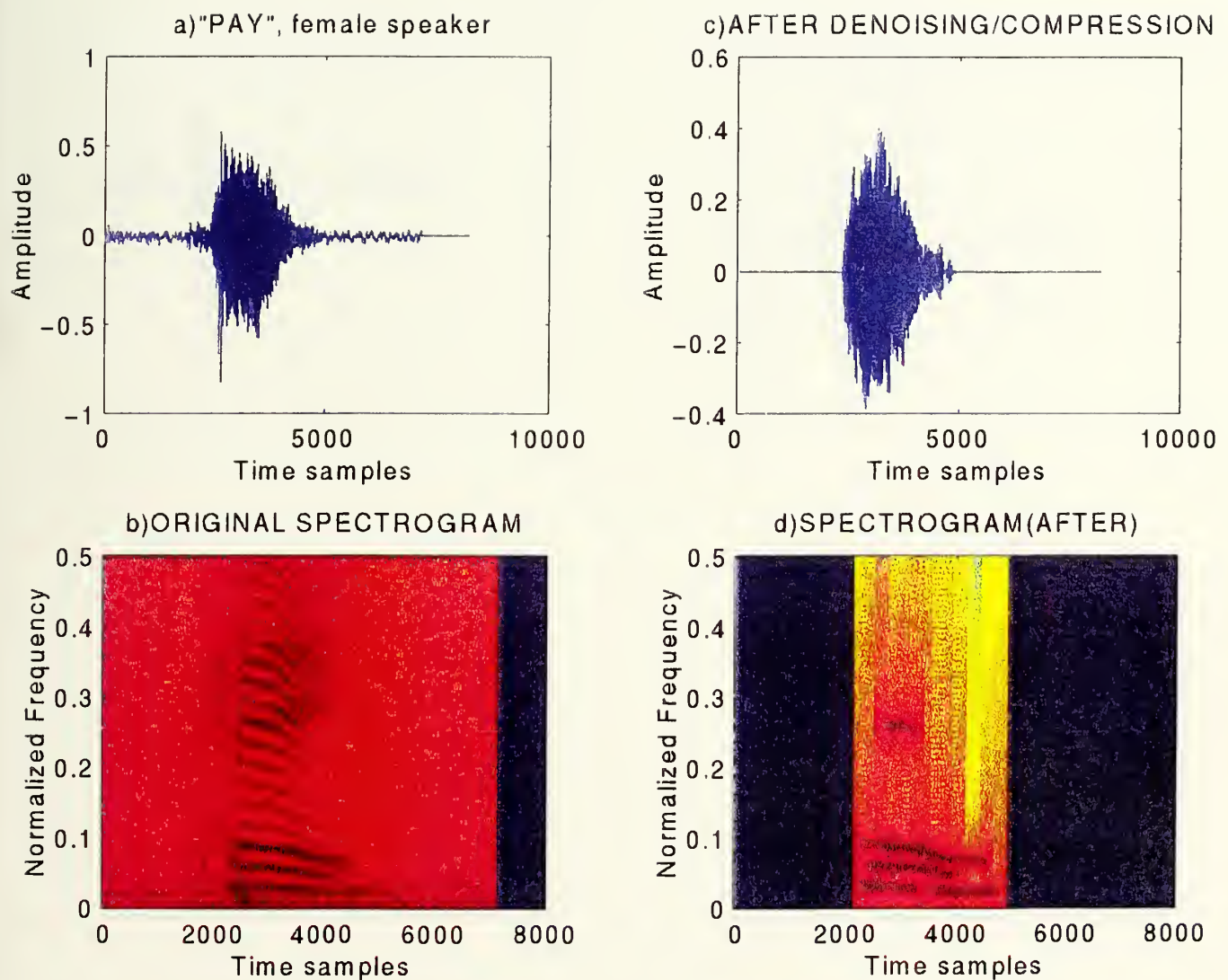


Figure 8.4 Word "Pay," female non-native speaker, "ndencomp" implementation;  
 (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot  
 after denoising/compression; (d) Spectrogram after denoising/compression (both  
 spectrograms use a Hanning time window of length 256 samples and overlapping of 128  
 samples between adjacent windows,  $f_s = 8$  KHz)



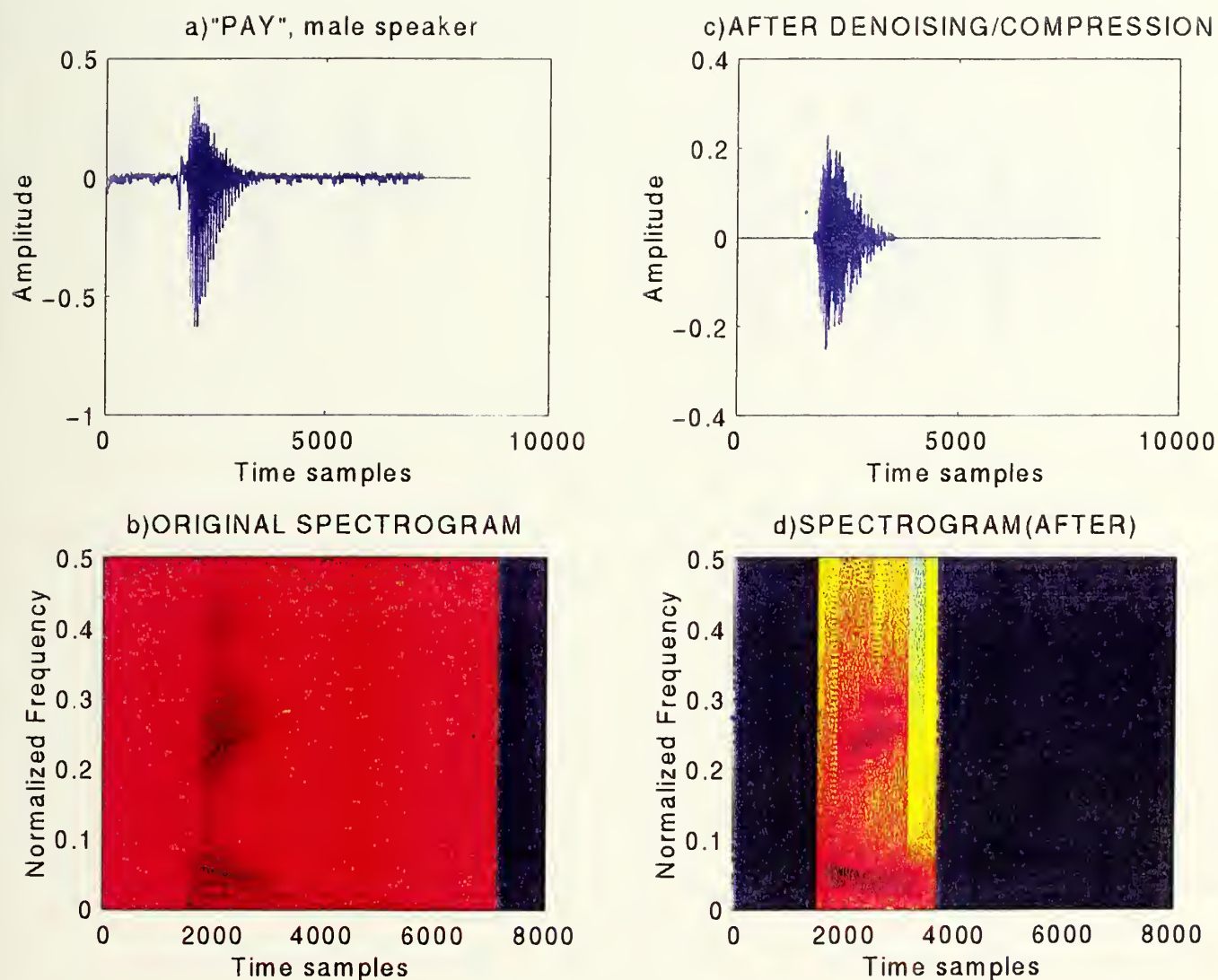


Figure 8.5 Word "Pay," male non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising/compression (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)





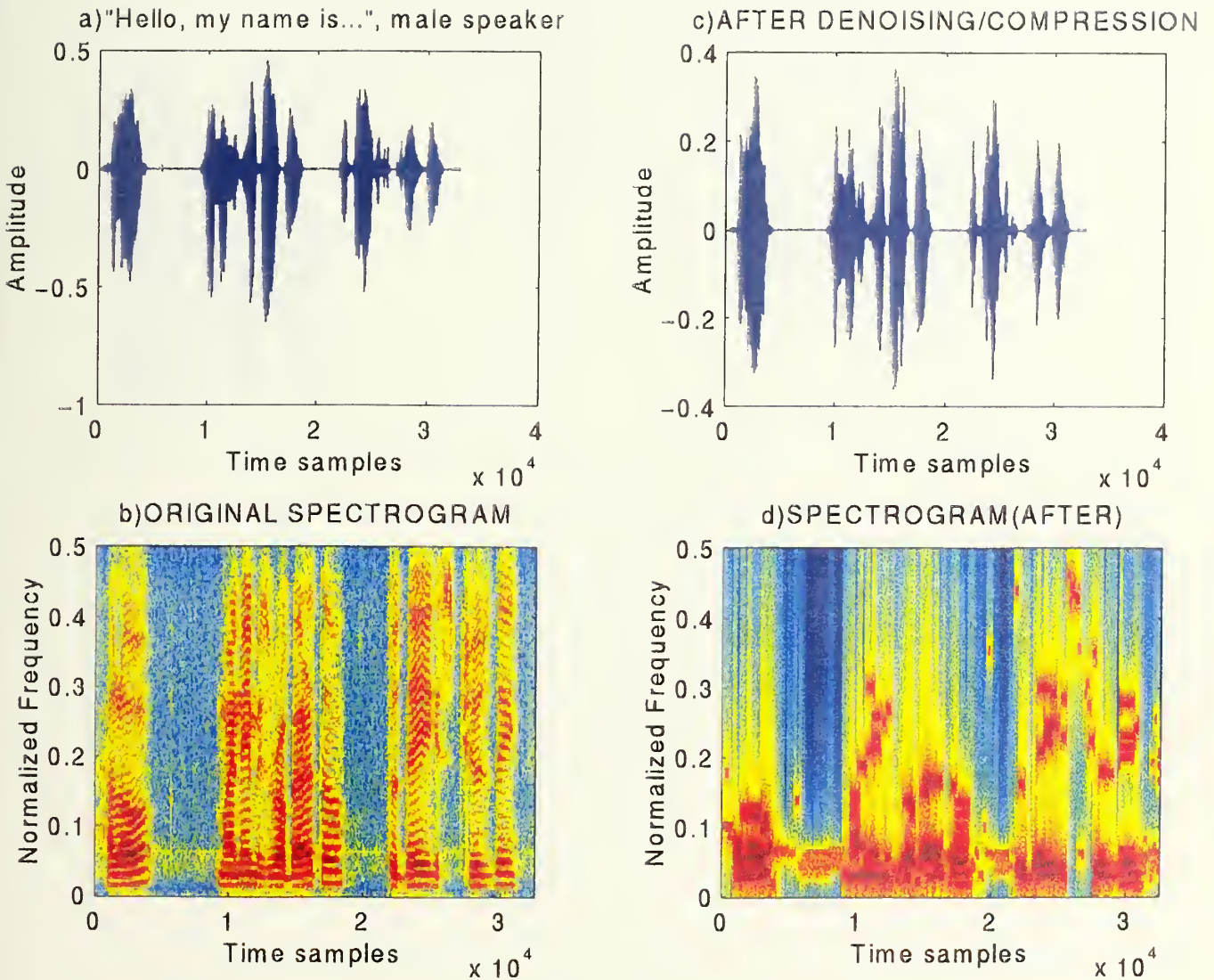


Figure 8.6 Sentence "Hello, my name is Roberto, today is Tuesday," male non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising/compression (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)



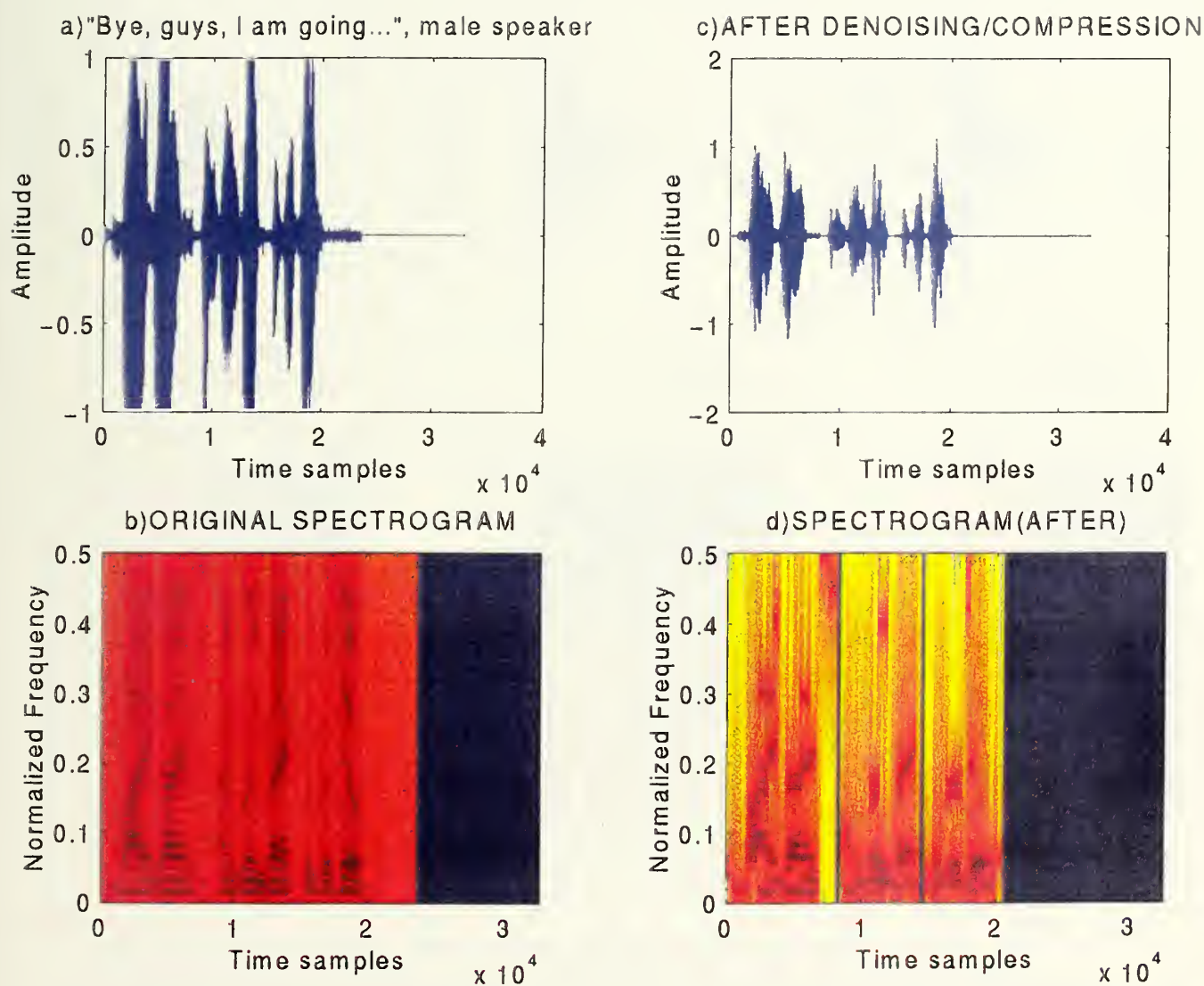


Figure 8.7 Sentence "Bye, guys, I'm going back to Brazil," male non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising/compression (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8\text{KHz}$ )





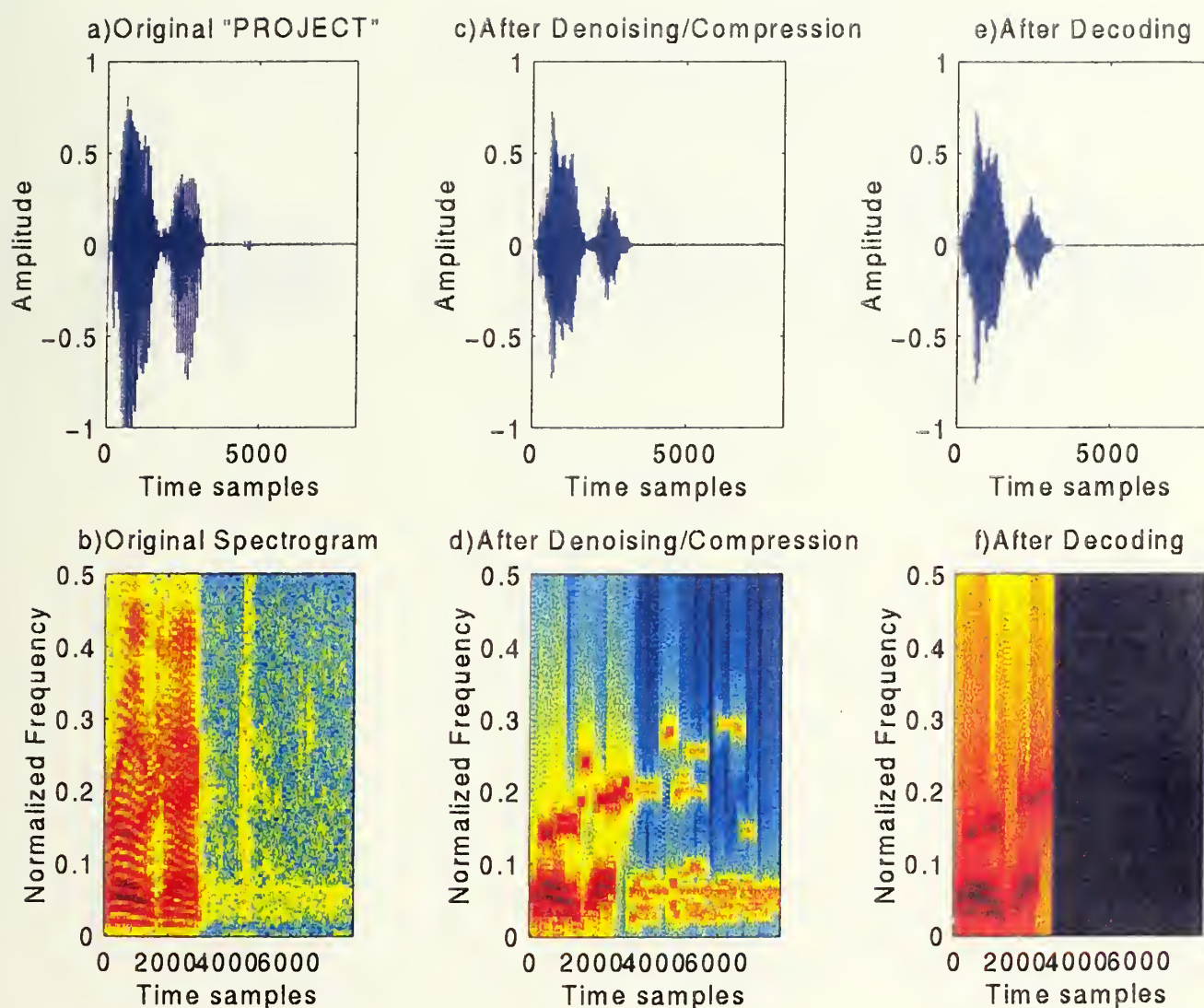


Figure 8.8 Word “*Project*,” male non-native speaker, “ndencomp” implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising/compression; (e) Time domain plot after decoding, 16-level quantizer; (f) Spectrogram after decoding, 16-level quantizer (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)





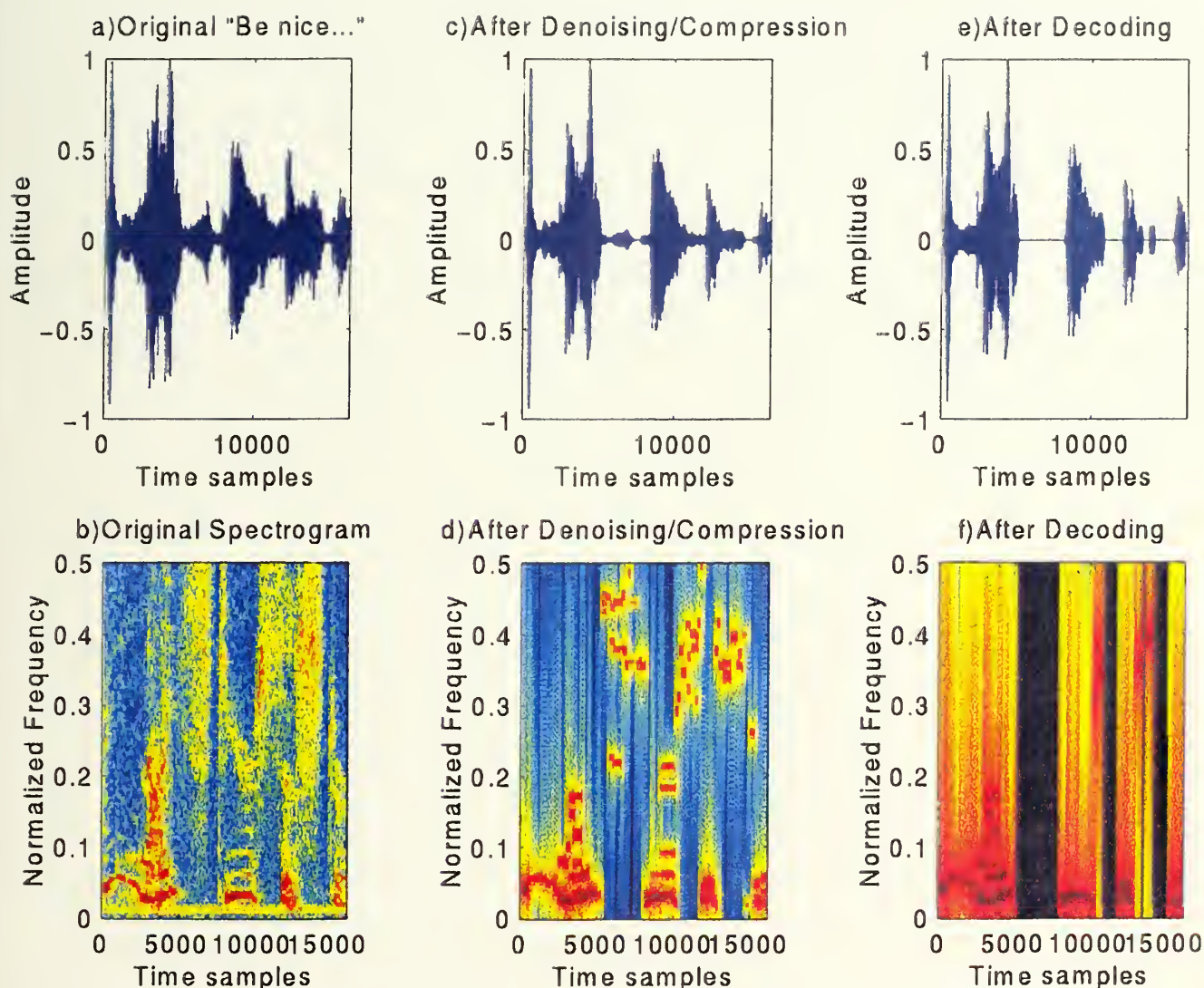


Figure 8.9 Sentence "Be nice to your sister," female native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising / compression; (e) Time domain plot after decoding, 16-level quantizer; (f) Spectrogram after decoding, 16-level quantizer (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)



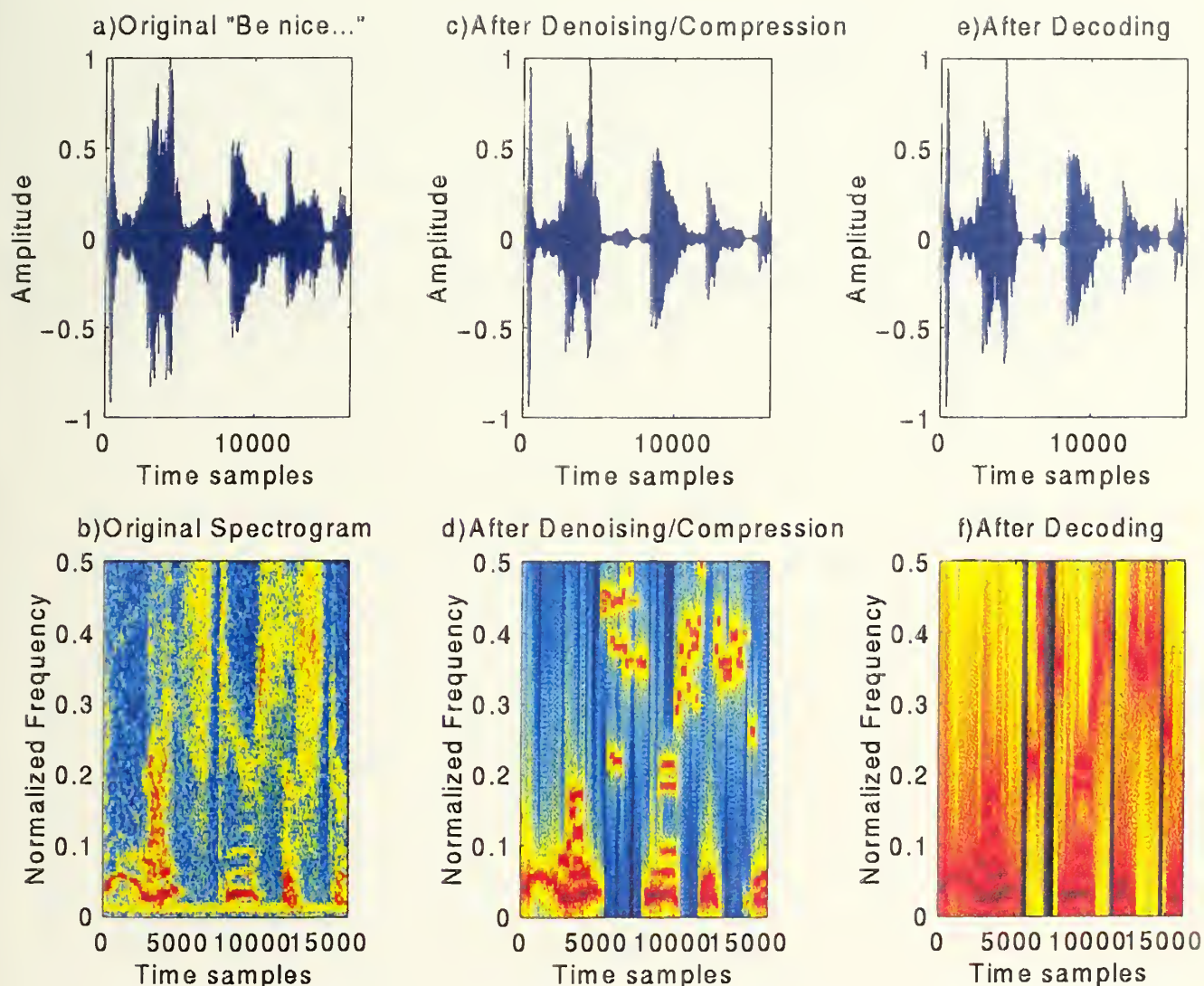


Figure 8.10 Sentence "Be nice to your sister," female native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising/compression; (e) Time domain plot after decoding, 32-level quantizer; (f) Spectrogram after decoding, 32-level quantizer (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)





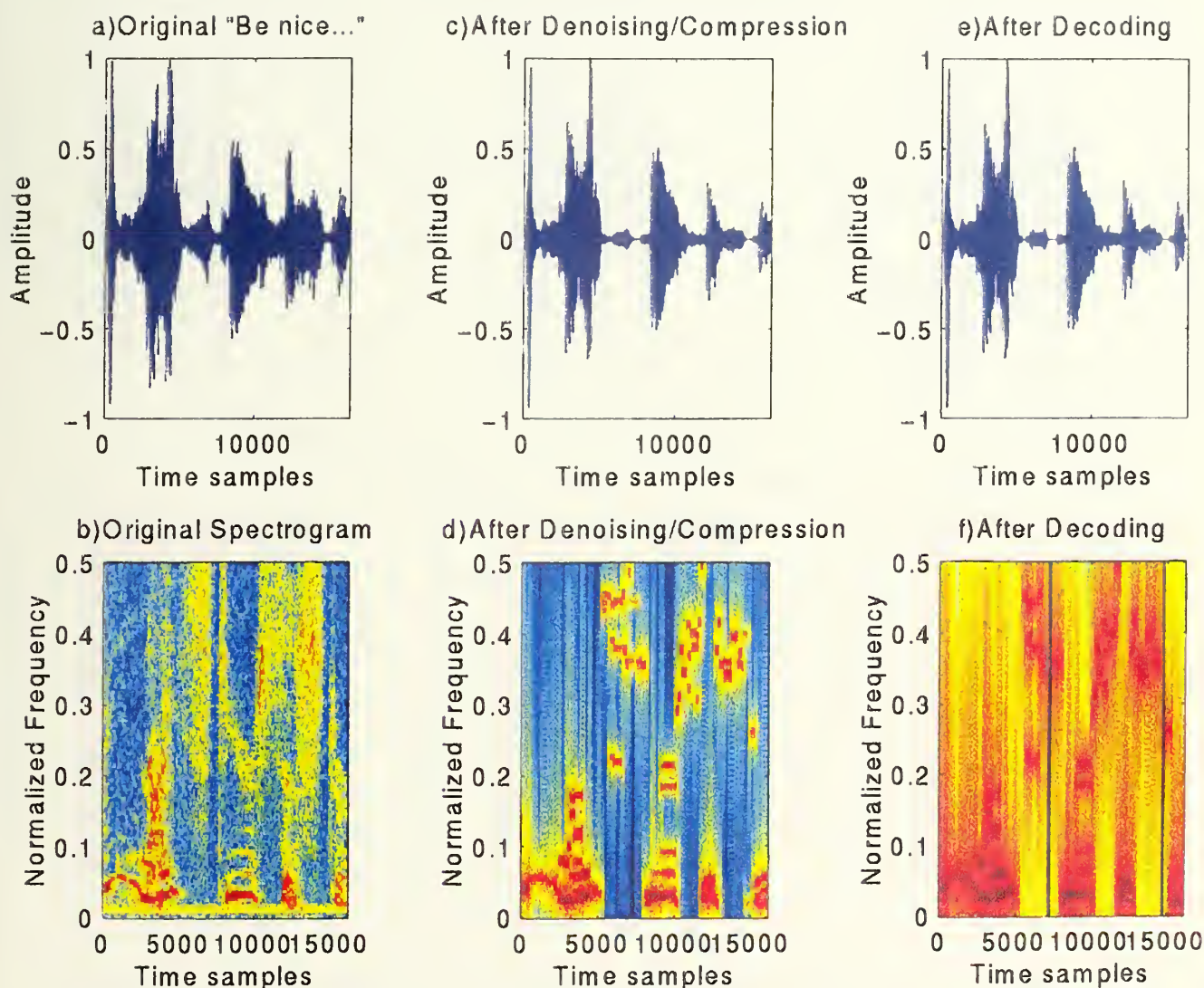


Figure 8.11 Sentence "*Be nice to your sister*," female native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Time domain plot after denoising/compression; (d) Spectrogram after denoising/compression; (e) After decoding, 64-level quantizer; (f) After decoding, 64-level quantizer (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)





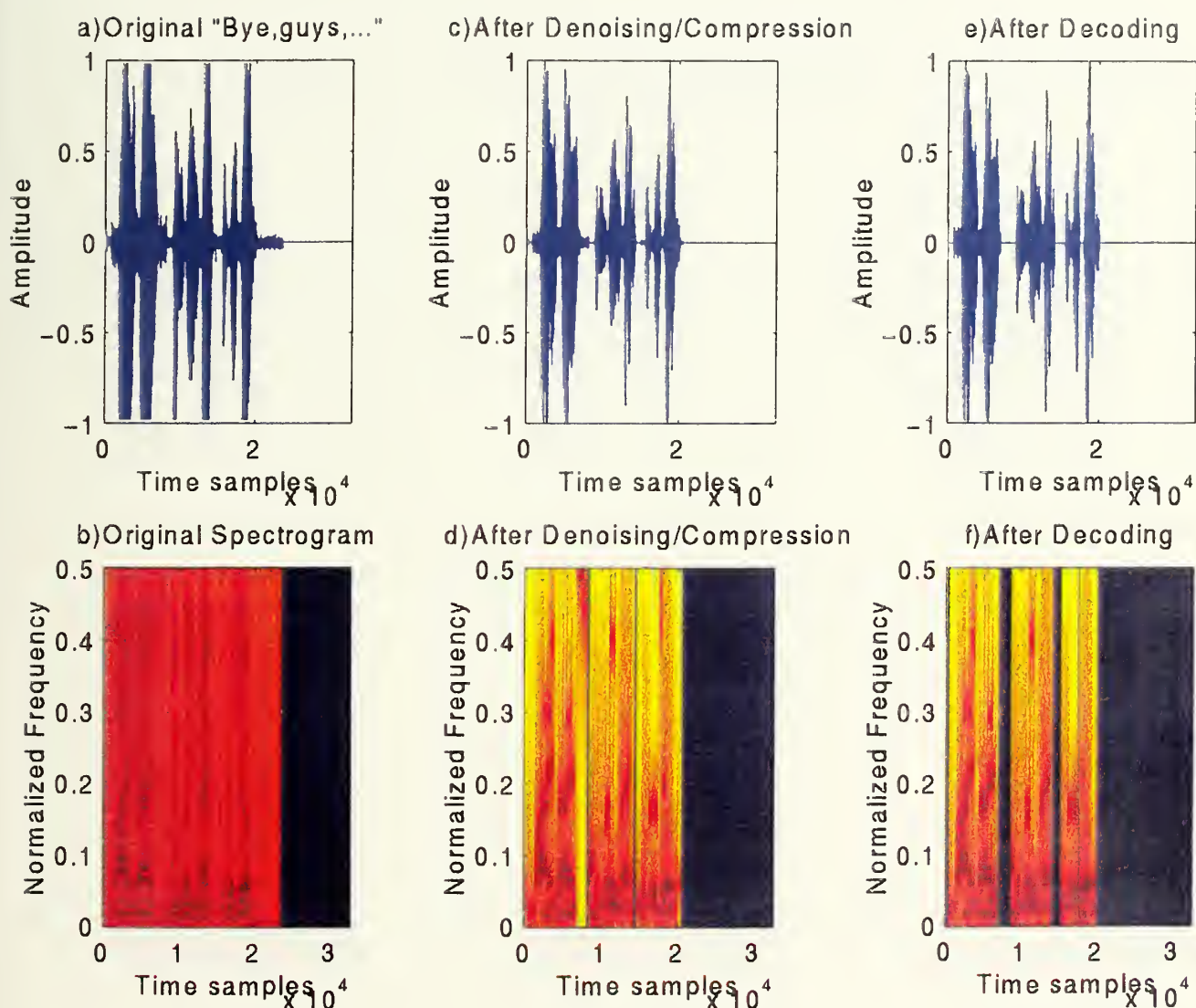


Figure 8.12 Sentence "Bye, guys, I'm going back to Brazil," male non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Plot after denoising/compression; (d) Spectrogram after denoising / compression; (e) Time domain plot after decoding, 16-level quantizer; (f) Spectrogram after decoding, 16-level quantizer (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)



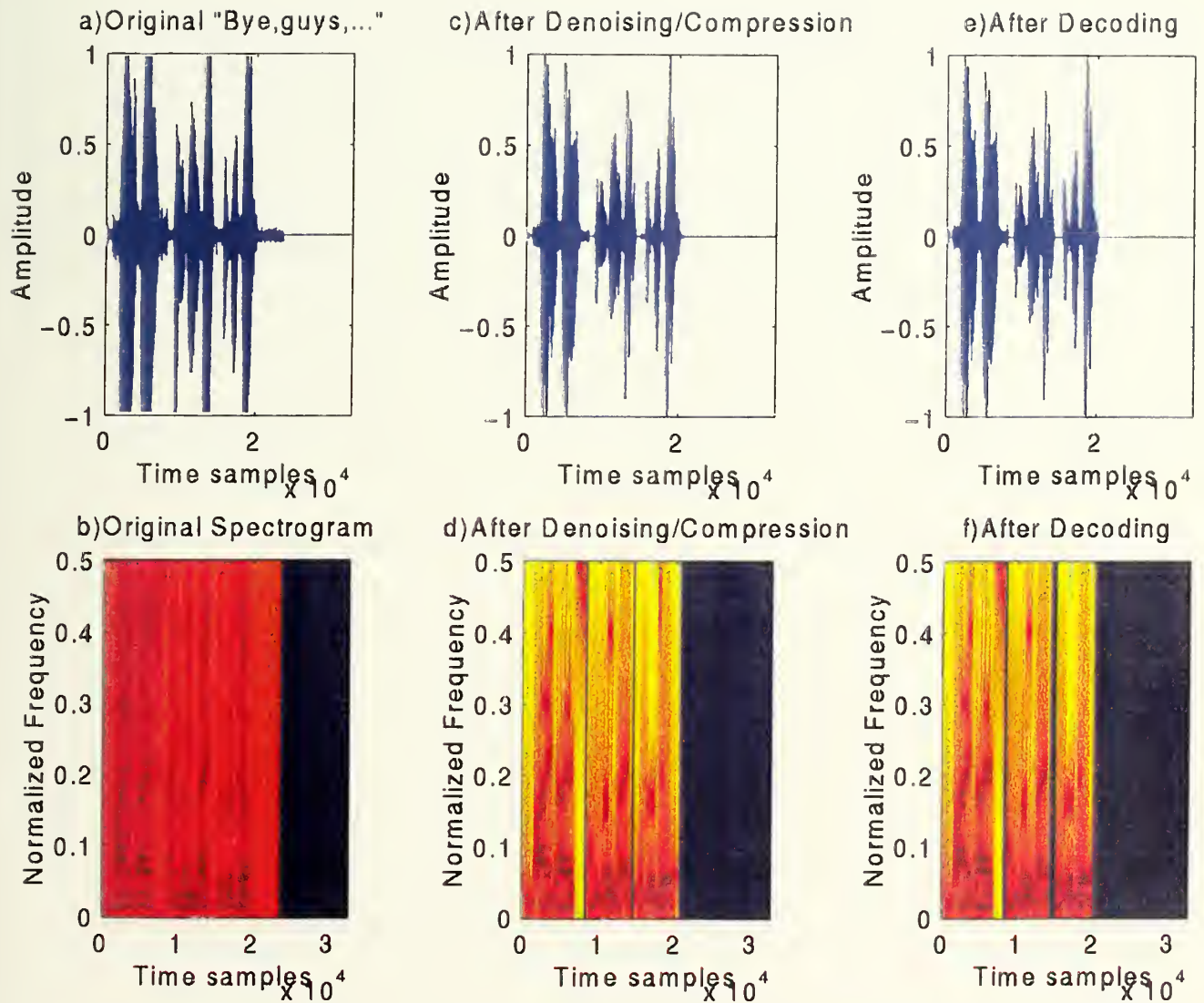


Figure 8.13 Sentence "Bye, guys, I'm going back to Brazil," male non-native speaker, "ndncomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Plot after denoising/compression; (d) Spectrogram after denoising / compression; (e) Time domain after decoding, 32-level quantizer; (f) Spectrogram after decoding, 32-level quantizer (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)



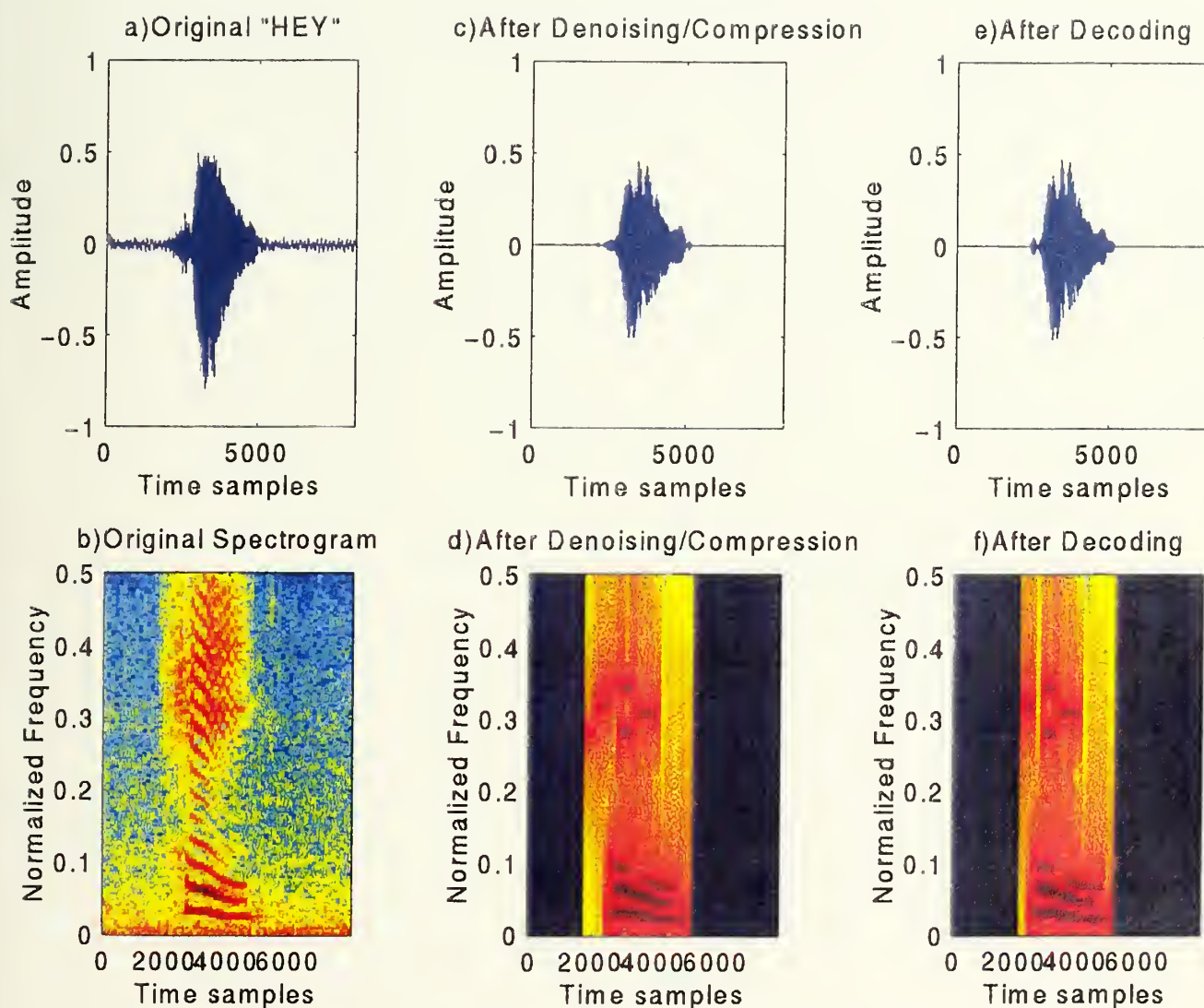


Figure 8.14 Word "Hey," female non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Plot after denoising/compression; (d) Spectrogram after denoising/compression; (e) Time domain plot after decoding, 16-level quantizer; (f) Spectrogram after decoding, 16-level quantizer (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8\text{KHz}$ )





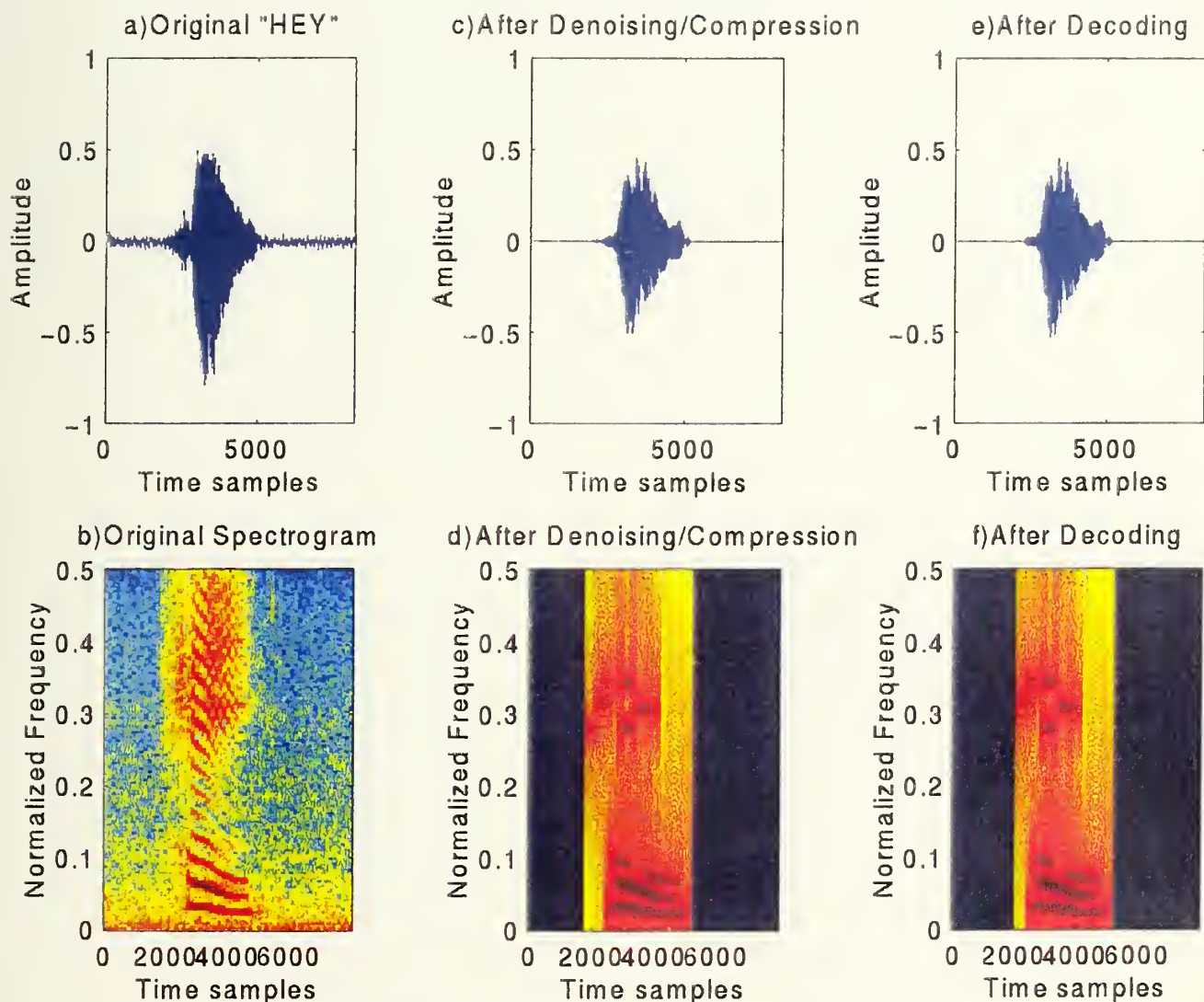


Figure 8.15 Word "Hey," female non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Plot after denoising/compression; (d) Spectrogram after denoising/compression; (e) After decoding, 32-level quantizer; (f) After decoding, 32-level quantizer (both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)



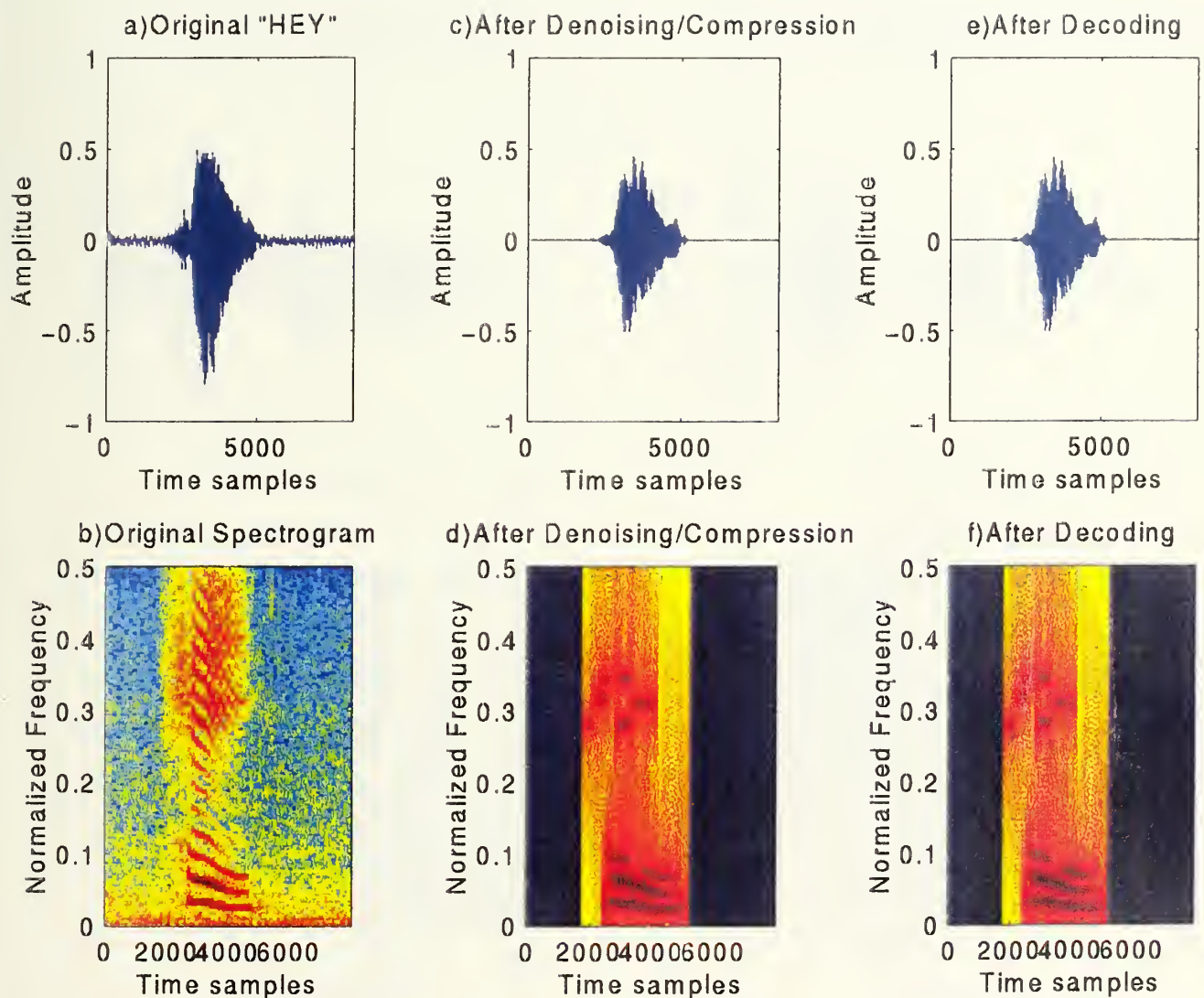


Figure 8.16 Word "Hey," female non-native speaker, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Plot after denoising /compression; (d) Spectrogram after denoising/compression; (e) After decoding, 64-level quantizer; (f) After decoding, 64-level quantizer ( both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)





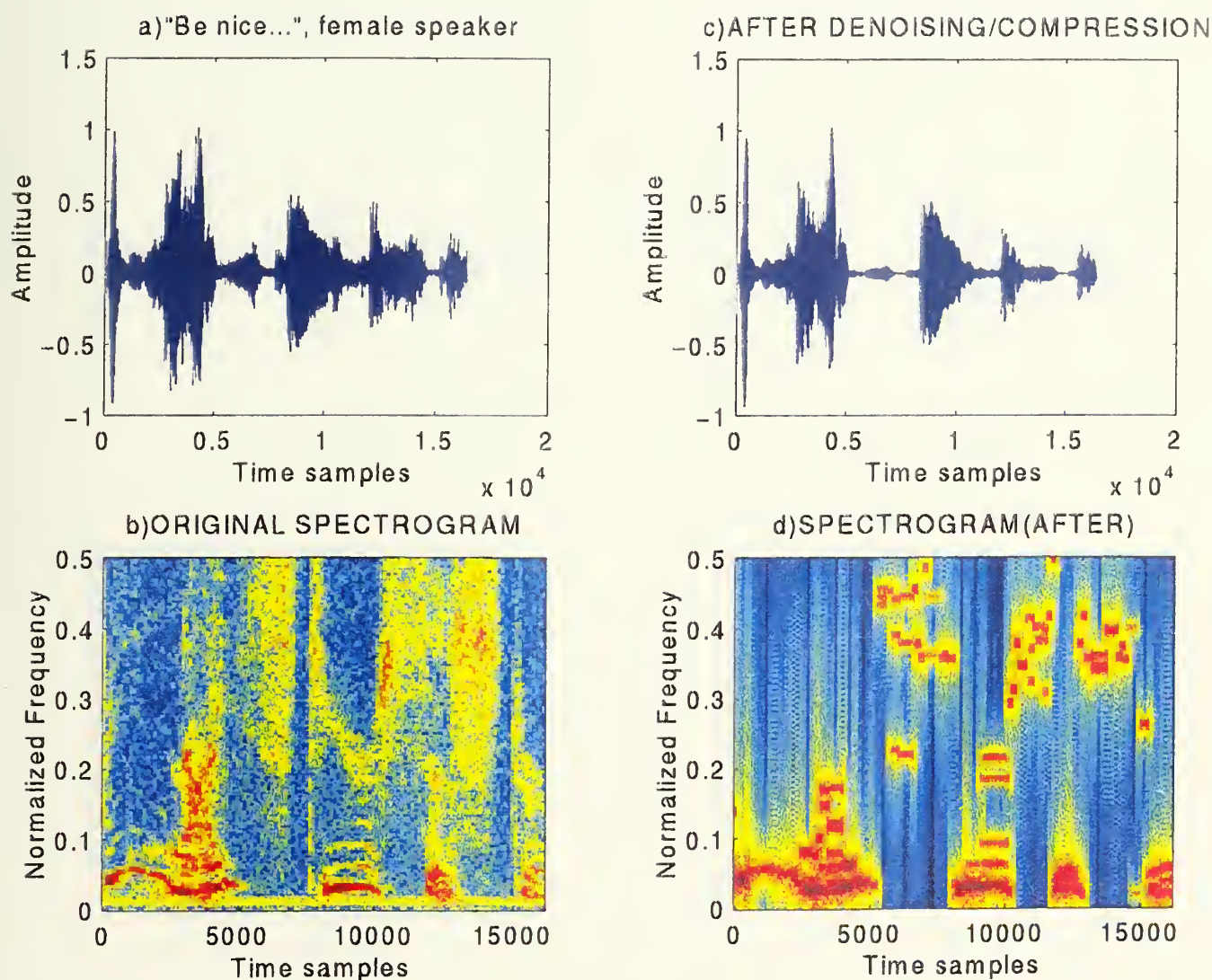


Figure 8.17 Sentence "Be nice to your sister," female native speaker, compressed with the CPT, "ndencomp" implementation; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Plot after denoising/compression with 0.85% non-zero coefficients selected; (d) Spectrogram after denoising/compression ( both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)





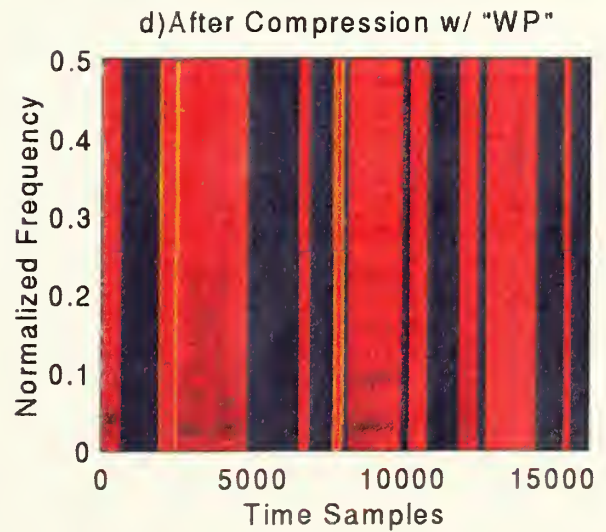
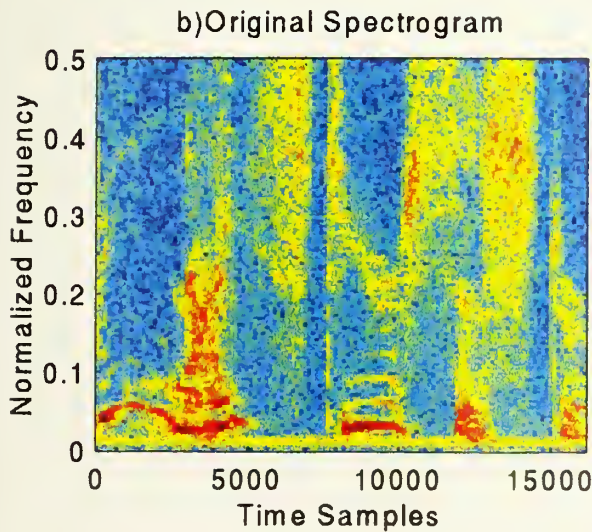
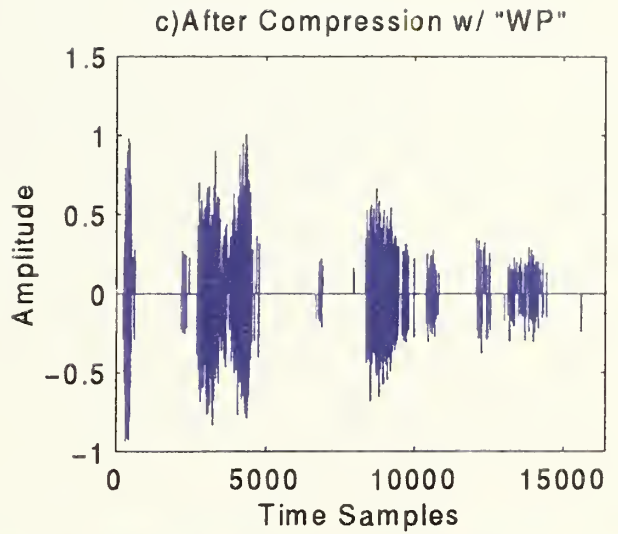
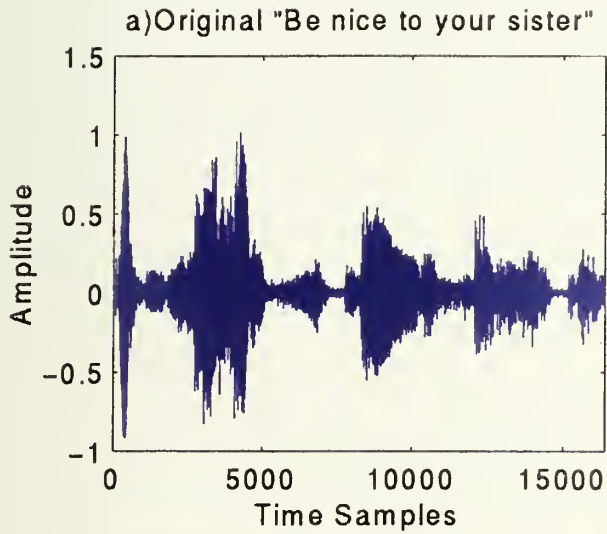


Figure 8.18 Sentence "*Be nice to your sister*," female native speaker, compressed with WPT, using a "Daubechies" basis function; (a) Original time domain plot; (b) Spectrogram of original speech; (c) Plot after denoising/compression with 15% non-zero coefficients selected; (d) Spectrogram after compression ( both spectrograms use a Hanning time window of length 256 samples and overlapping of 128 samples between adjacent windows,  $f_s = 8$  KHz)







## IX. CONCLUSION

In this thesis, compression schemes based on the Cosine Packet Transform using the Local Cosine Transform are presented. The basis functions are chosen via the Best Basis Algorithm using the entropy minimization criterion.

Coefficients for compression are chosen with an adaptive scheme, which selects more cosine packet coefficients for voiced intervals than for unvoiced ones. In addition, since some recorded speech sounds have equipment noise, a denoising scheme is performed.

Finally, an encoding scheme is implemented. Thus, this study simulates the entire process of denoising, compression, and encoding (on the transmitter side), as well as decoding and reconstruction (on the receiver side).

The results obtained are good, due to the combination of certain factors, which include the following:

- (a) Good time and frequency resolution of the local cosine transform;
- (b) The Cosine Packet Transform, combined with the Best Basis algorithm using the entropy minimization criterion allowed not only for minimizing the entropy, but also for the splitting of the signal into its locally stationary portions. These two factors greatly contribute to the success of the compression scheme;
- (c) The Adaptive Thresholding scheme helps to optimize in quality and quantity the number of cosine packet coefficients, while preserving good compressed signal properties;
- (d) The denoising scheme allows the number of non-zero coefficients to be reduced and, at the same time, a better quality of denoised sound when compared to the original noisy speech.

Through the denoising attempts, it is possible to recognize some patterns of speech that would be hidden by the higher energy noise in regular compression. The



frequency analysis allows differentiating speech sounds and background noise and hence permits recovering most of the speech sounds.

Basically, two main problems remain. First, there are a few sounds with low energy that need to be correctly identified and recovered from the background noise. In the experiments with noisy speech, the only cases that could not be solved are the weak unvoiced endings, like /t/ at the the word “met,” (which is reconstructed like a /d/) and /s/ at the end of words “cats” and “lets”, which is lost due to the denoising process. Although many phonemes were tried, there are probably some others that could have been attempted and, thus, this is a suggestion for further study. The second problem that was encountered is quantization noise. Although the encoding scheme works well enough to make speech recognition for many cases in the simulated receiver sounding “cleaner” than the original noisy signal, noise is introduced by the quantization process. Although very small, this noise is enough for cancelling endings like /kt/ in the word “project”. Since this research focused on the compression schemes, less effort is made to develop a better quantizing and encoding schemes (another point for further study).

The CPT performs better than the WPT for speech compression applications. When using the WPT, the compression scheme begins losing low energy sounds much earlier than the CPT, i.e., with a much lower compression ratio, although this may be due to the basis function that was selected.

The purpose of this study is to find an optimal scheme for the compression of speech signals. Since the scheme used in this study is successful, speech samples with the highest possible compression ratios are tested. The quality reconstruction that results for the majority of tries can be considered as “fair” (see Table 8.1), as shown by the average mean grades assigned. The very small percentages of selected coefficients in the compression scheme result tables, and very high compression ratios for the encoding results, together with a “fair” quality reconstruction indicate a positive overall result. The compression ratios are not fixed, since the scheme is adaptive to the speech being analyzed. However our results indicate an average compression ratio of 1:50 on the

speech used in our study. The ratio can be adjusted for better quality of reconstructed sound, according to the needs and availability of the user. Evidently, there will always be a need to compromise between the compression ratio and the quality of the reconstructed speech.



## APPENDIX. COMPUTER CODE

```
% Name: Compcp.m and necompcp.m
% Subject: Analysis, Compression and Synthesis routine of speech data
% Description:
% These two routines contain the following main parts:

% a) Input and loading of speech to be used ( prompts the user for choices like gender of
% speaker, word or sentences among those available and finest depth for time splitting);

% b) Implements the Cosine Packet Transform (CPT) of the speech sequence;

% c) Chooses the basis for the CPT by applying the Best Basis Algorithm;

% d) Implements a Frequency Behavior and an Energy Behavior plot;

% e) Implements a voiced-unvoiced segmentation;

% f) Selects the coefficients by applying the Adaptive Thresholding scheme;

% g) Applies the inverse CPT, by transforming each interval, unfolding and adding it to the
% existing sequence;

% h) Computes and presents the number of non-zero coefficients before and after the
% compression scheme as well as the mean square error between the original and the
% reconstructed sequences;

% i) Presents plots containing the Frequency as well as the Energy behavior; also presents the
% voiced-unvoiced segmentation plot as well as time domain and spectrogram plots of both
% original and reconstructed sequences;

% Note1: Parts b), c) and g) are extracted from the software package Wavelab.600, Stanford
% University[17]. This is also valid for the programs encp6.m, ndencomp.m,
% encptour.m and ndentour.m;
% Note 2: WaveLab code was modified to implement our compression schemes.
% Written and adapted by J. Roberto V. Martins, in October 1995.

% Compcp.m

% Input and loading of speech to be used

clear;
V = input('Please enter "1" for female voice and "2" for a male voice : ');
if V==1
    P = 2;
    FV = input('Please enter 1 for the sentence, 2 for "be" , 3 for "hate", 4 for "hey" , 5 for "met" , 6 for
"pay", 7 for "cats", 8 for "benice" : ');
    if FV == 1
        clear ny;
```

```

load fse;
ny = [fse' zeros(1,5120)];

elseif FV==2
clear ny;
load fbe;
ny = [fbe' zeros(1,2048)];
elseif FV==3
clear ny;
load fha;
ny = [fha' zeros(1,7168)];
elseif FV==4
clear ny;
load fhey;
ny = [fhey'];
elseif FV==5
clear ny;
load fmet;
ny = [fmet' zeros(1,1024)];
elseif FV==6
clear ny;
load fpay;
ny = [fpay' zeros(1,1024)];
elseif FV==7
clear ny;
load fcats;
ny = [fcats'];
elseif FV==8
clear ny;
benice = loadwav('benice.wav');
ny = [(benice(1:16384)/max(abs(benice))+0.0119)'];
end
end
if V==2
P = 2;
W = input('Please enter 1,for "project",2 for "cataratas",3 for "encyclopedia", 4 for "issos",5 for "assos",6
for "six",7 for "the sentence",8 for "aka",9 for "at",10 for "azure",11 for "be",12 for "bird",13 for "boot",14
for "call",15 for "day",16 for "eka",17 for "epa", 18 for "eve",19 for "father",20 for "foot", 21 for "for", 22
for "go", 23 for "hate", 24 for "he",25 for "ika",26 for "it",27 for "key",28 for "let",29 for "me",30 for
"met",31 for "no",32 for "obey",33 for "opa",34 for "pay",35 for "read",36 for "see",37 for "she",38 for
"then",39 for "thin", 40 for "to",41 for "up", 43 for "vote",44 for "we", 45 for "you", 46 for "zoo",47 for
"silence", 48 for "the bye sentence",49 for "beback", 50 for "blows", 51 for "bruna",52 for "adams", 53 for
"sounds good" : ');

if W==1
clear ny;
load newvoice;
ny = y(2700:2700+8191)';
elseif W==2
clear ny;
load catar;
ny = ca(1900:1900+8191)';
elseif W==3

```

```

clear ny;
load encic;
ny = en(1200:1200 +8191)';
elseif W==4
clear ny;
load issos
ny = is(1900:1900+8191)';
elseif W==5
clear ny
load assos
ny = as(1900:1900+8191)';
elseif W==6
clear ny
load six
ny = si(1:8192)';
elseif W==7
clear ny;
load myvoice;
ny = x(9000:9000+32767)';
elseif W==8
clear ny;
load aka;
ny = (ac+0.1656)';
elseif W==9
clear ny;
load at;
ny = (at+0.1655)';
elseif W==10
clear ny;
load azure;
ny = [(az+0.1651)' zeros(1,6144) ];
elseif W==11
clear ny;
load be;
ny = [(be+0.1654)' zeros(1,3072) ];
elseif W==12
clear ny;
load bird;
ny = [(bi+0.1658)' zeros(1,7168) ];
elseif W==13
clear ny;
load boot;
ny = [(bo+0.1652)'];
elseif W==14
clear ny;
load call;
ny = [(cal+0.1654)' zeros(1,6144) ];
elseif W==15
clear ny;
load day;
ny = [(da+0.1645)' zeros(1,1024) ];
elseif W==16
clear ny;
load eka;

```



```

    ny = [(ek+0.1653)'];
elseif W==17
    clear ny;
    load epa;
    ny = [(ep+0.1650)' zeros(1,6144) ];
elseif W==18
    clear ny;
    load eve;
    ny = [(ev+0.1654)' zeros(1,4096) ];
elseif W==19
    clear ny;
    load father;
    ny = [(fa+0.1648)' zeros(1,6144) ];
elseif W==20
    clear ny;
    load foot;
    ny = [(foo+0.1653)' zeros(1,6144) ];
elseif W==21
    clear ny;
    load for;
    ny = [(fo+0.1649)' zeros(1,6144) ];
elseif W==22
    clear ny;
    load go;
    ny = [(go+0.1651)'];
elseif W==23
    clear ny;
    load hate;
    ny = [(ha+0.1657)' zeros(1,7168) ];
elseif W==24
    clear ny;
    load he
    ny = [(he+0.1657)' zeros(1,2048) ];
elseif W==25
    clear ny;
    load ika
    ny = [(ik+0.1654)' zeros(1,6144) ];
elseif W==26
    clear ny;
    load it
    ny = [(it+0.1657)' zeros(1,3072) ];
elseif W==27
    clear ny;
    load key;
    ny = [(ke + 0.1652)' zeros(1,2048)];
elseif W==28
    clear ny;
    load let;
    ny = [(le + 0.1657)'];
elseif W==30
    clear ny;
    load met;
    ny = [(met + 0.1653)'];
elseif W==31

```

```

clear ny;
load no;
ny = [(no + 0.1646)' zeros(1,1024)];
elseif W==34
clear ny;
load pay;
ny = [(pa + 0.1655)' zeros(1,1024)];
elseif W==36
load see;
ny = [(se+0.1653)' zeros(1,1024)];
elseif W==37
load she;
ny = [(sh+0.1654)'];
elseif W==38
load then;
ny = [(th+0.1656)' zeros(1,6144)];
elseif W==39
load thin;
ny = [(thi + 0.1655)' zeros(1,1024)];
elseif W==40
load to;
ny = [(to + 0.1649)' zeros(1,3072)];
elseif W==41
load up;
ny = [(up + 0.1653)' zeros(1,2048)];
elseif W==43
load vote;
ny = [(vo + 0.1654)' zeros(1,1024)];
elseif W==44
clear ny;
load we
ny = [(we+0.1655)' zeros(1,2048) ];
elseif W==45
clear ny;
load you
ny = [(you + 0.1655)' zeros(1,2048) ];
elseif W==46
clear ny;
load zoo
ny = [(zo+0.1646)' zeros(1,2048) ];
elseif W ==47
clear ny;
load myvoice;
ny = x(1:8192)';
elseif W ==48
clear ny;
load bye;
ny = [ bye' zeros(1,9216)];
elseif W ==49
clear ny;
beback = loadwav('beback.wav');
ny = [ (beback/max(abs(beback))+0.056)' zeros(1,4824)];
elseif W ==50
clear ny;

```

```

blows = loadwav('blows.wav');
ny = [ (blows(1:16384)/max(abs(blows))+0.0034)'];
elseif W==51
clear ny;
br = loadwav('bruna.wav');
ny = [ (br/max(abs(br)) + 0.0155)' zeros(1,7268) ];
elseif W==52
clear ny;
adam = loadwav('adamsfam.wav');
ny = [ (adam(1:32768)/max(abs(adam)) + 0.0081)'];
elseif W==53
clear ny;
load engl6;
ny = [ (engl6(1:16384) + 3.019e-4)'];

end

end

n = length(ny)
D = input('Enter the finest depth for Time Splitting : ');

% Implementing the Cosine Packet Transform

cp = CPAnalysis(ny,D,'Sine');
stree = CalcStatTree(cp,'Entropy');
[btree,vtree] = BestBasis(stree,D);
[n,L] = size(cp);
% Create Bell

bellname = 'Sine';
m = n / 2^D / 2;
[bp,bm] = MakeONBell(bellname,m);
x = zeros(1,n);

% initialize tree traversal stack

stack = zeros(2,2^D+1);
tp = zeros(1,n);
v = zeros(1,n);
compr = zeros(1,n);
coef = zeros(1,n);
ncoef = zeros(1,n);
k = 1;
stack(:,k) = [0 0]';
v = zeros(1:n);
vs = zeros(1:n);
ind = 0;
le = zeros(1,2^D);
while(k > 0),
d = stack(1,k); b = stack(2,k); k=k-1;
if(btree(node(d,b)) ~= 0) , % nonterminal node
k = k+1; stack(:,k) = [(d+1) (2*b) ]';

```

```

        k = k+1; stack(:,k) = [(d+1) (2*b+1)];
    else
        c = cp(packet(d,b,n),d+1)';
        coef(1,b/(2^d).*n+1:(b+1)/(2^d).*n) = c;
        i = (b/(2^d).*n+1);
        len = length(c);
        [ I,ND] = max(abs(c));
        compr(1,b/(2^d).*n+1) = length(c);
    end

```

% Identifying the Frequency Content of each interval

```

    if ND <= round(len/16)

        v(i) = 0.25; % it was 0.2

    elseif ND <= round(len/8)

        v(i) = 0.5; % it was 0.4

    elseif ND < length(c)/(2*P)

        v(i) = 1; % it was 0.6
    else
        [sI,sND] = max(abs([coef(i:i+ND-3),0,0,coef(i+ND+1:i+len-1)]));
        if sND >= length(c)/(2*P)
            if ND <= len/2
                v(i) = 1.5; %it was 0.75;
            elseif ND <= len*3/4
                v(i) = 2; % it was 0.9
            else
                v(i) = 2.5; % it was 1.0
            end
        elseif sND > round(len/8)
            v(i) = 1; % it was 0.6

        elseif sND > round(len/16)
            v(i) = 0.5; % it was 0.4
        else
            v(i) = 0.25; % it was 0.2
        end
    end

    end

    ec(i:i+len-1) = ones(1,len) .* sum(c.^2); % computing the energy of the coefficients
    es(i:i+len-1) = ones(1,len) .* sum(ny(i:i+len-1).^2); % computing the energy of the intervals
    vari = std(c);

    tp (1,b/(2.^d).*n+1) = 1;
    len = length(c);
    ind = ind +1;
    le(ind) = log2(len);
    v(i);
    rko = length(c)/16;
    ko = ND;

```

```

fo = 4000/length(c)*ND;
toten = sum(coef.^2);
i = (b/(2^d)*n+1);

```

% Applying the Adaptive Thresholding Compression Scheme

```

if v(i) <= 0.5

    if sum(coef(i:compr(i)-1+i).^2) < toten/n * len
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);
        else
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);
        end
    else
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),98.7);
        else
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),97.66);
        end
    end

    nc = nncoef(i:compr(i)-1+i);

end

if v(i) > 0.5
    sumco = sum(coef(i:compr(i)-1+i).^2);
    thres = 0.5*toten/n * len;
    if sum(coef(i:compr(i)-1+i).^2) < toten/n * len;
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);
        else
            nncoef(i:len+i-1) = comp(coef(i:len+i-1),99.5);
        end
    else
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);
        else
            nncoef(i:compr(i)-1+i) = comp(coef(i:compr(i)-1+i),99.5);
        end
    end

    nc = nncoef(i:compr(i)-1+i);

end

if v(i) > 1
    vs(i) = 1;
else
    if es(i:i+len-1) > (toten/n*2.5*len)
        vs(i) = 0.5;
    end
end

```

```

        end
    end
    y = dct_iv(nc); % Inverse Transforming each interval

% Unfolding each interval and Reconstructing the time sequence after compression

    [xc,xl,xr] = unfold(y,bp,bm);
    x(packet(d,b,n)) = x(packet(d,b,n)) + xc;
    if b>0,
        x(packet(d,b-1,n)) = x(packet(d,b-1,n)) + xl;
    else
        x(packet(d,0,n)) = x(packet(d,0,n)) + edgeunfold('left',xc,bp,bm);
    end
    if b < 2^d-1,
        x(packet(d,b+1,n)) = x(packet(d,b+1,n)) + xr;
    else
        x(packet(d,b,n)) = x(packet(d,b,n)) + edgeunfold('right',xc,bp,bm);
    end
end

end
nind = sum(le>0);
nle = le(1:nind);
figure(1),plot(ny) , hold;
plot(tp,':'),hold off;
figure(2),plot(ny),hold
plot(v,':'),hold off;
mse = mean((ny - x).^2) % computing the mean square error between the original and
    % the reconstructed sequence;

scoefm0 = sum(abs(coef)>0) % computing the number of non-zero coefficients before
    % compression
sncoefm0 = sum(abs(nncoef)>0) % computing the number of non-zero coefficients after
    % compression

figure(3),
plot(x);
figure(4),
plot(ec);
figure(5),
plot(es);
figure(6),specgram(ny,[],1)
title('Observing the Coarticulation for the sound "ISSOS"')
print figure6-depsc
figure(7),
subplot(3,1,1),plot(ny)
title('Speech Signal: "ISSOS"')
subplot(3,1,2),plot(ny) , hold;
plot(tp,':'),hold off;title('Time Partition')
subplot(3,1,3),plot(ny),hold
plot(v,':'),hold off;title('Frequency Behavior')
figure(8),
subplot(3,1,1),plot(ny,'b'),
%plot(v,':'),hold off;

```



```

title(' "Be nice to your sister"')
subplot(3,1,2),plot(vs,'b')
title('Voiced-Unvoiced Segmentation')
subplot(3,1,3),
specgram(ny,[],1)
title('Observing The Spectrogram for "Be nice to your sister"')
print figure7 -depsc

```

```
% Necompcp.m
```

```
% Input and loading of speech to be used
```

```
clear;
```

```
V = input('Please enter "1" for female voice and "2" for a male voice : ');
```

```
if V==1
```

```
    P = 2;
```

```
    FV = input('Please enter 1 for the sentence, 2 for "be" , 3 for "hate", 4 for "hey" , 5 for "met" , 6 for "pay", 7 for "cats", 8 for "benice" : ');
```

```
    if FV == 1
```

```
        clear ny;
```

```
        load fse;
```

```
        ny = [fse' zeros(1,5120)];
```

```
    elseif FV==2
```

```
        clear ny;
```

```
        load fbe;
```

```
        ny = [fbe' zeros(1,2048)];
```

```
    elseif FV==3
```

```
        clear ny;
```

```
        load fha;
```

```
        ny = [fha' zeros(1,7168)];
```

```
    elseif FV==4
```

```
        clear ny;
```

```
        load fhey;
```

```
        ny = [fhey'];
```

```
    elseif FV==5
```

```
        clear ny
```

```
        load fmet;
```

```
        ny = [fmet' zeros(1,1024)];
```

```
    elseif FV==6
```

```
        clear ny
```

```
        load fpay
```

```
        ny = [fpay' zeros(1,1024)];
```

```
    elseif FV==7
```

```
        clear ny
```

```
        load fcats
```

```
        ny = [ fcats'];
```

```
    elseif FV==8
```

```
        clear ny
```

```
        benice = loadwav('benice.wav');
```

```
        ny = [(benice(1:16384)/max(abs(benice))+0.0119)'];
```

```
    end
```

```
end
```

```
if V==2
```

```

P = 2;
W = input('Please enter 1,for "project",2 for "cataratas",3 for "encyclopedia", 4 for "issos",5 for "assos",6
for "six",7 for "the sentence",8 for "aka",9 for "at",10 for "azure",11 for "be",12 for "bird",13 for "boot",14
for "call",15 for "day",16 for "eka",17 for "epa", 18 for "eve",19 for "father",20 for "foot", 21 for "for", 22
for "go", 23 for "hate", 24 for "he",25 for "ika",26 for "it",27 for "key",28 for "let",29 for "me",30 for
"met",31 for "no",32 for "obey",33 for "opa",34 for "pay",35 for "read",36 for "see",37 for "she",38 for
"then",39 for "thin", 40 for "to",41 for "up", 43 for "vote",44 for "we", 45 for "you", 46 for "zoo",47 for
"silence", 48 for "the bye sentence",49 for "beback", 50 for "blows", 51 for "bruna",52 for "adams" : ');

if W==1
    clear ny;
    load newvoice;
    ny = y(2700:2700+8191)';
elseif W==2
    clear ny;
    load catar;
    ny = ca(1900:1900+8191)';
elseif W==3
    clear ny;
    load encic;
    ny = en(1200:1200 +8191)';
elseif W==4
    clear ny;
    load issos
    ny = is(1900:1900+8191)';
elseif W==5
    clear ny
    load assos
    ny = as(1900:1900+8191)';
elseif W==6
    clear ny
    load six
    ny = si(1:8192)';
elseif W==7
    clear ny;
    load myvoice;
    ny = x(9000:9000+32767)';
elseif W==8
    clear ny;

    load aka;
    ny = (ac+0.1656)';
elseif W==9
    clear ny;
    load at;
    ny = (at+0.1655)';
elseif W==10
    clear ny;
    load azure;
    ny = [(az+0.1651)' zeros(1,6144) ];
elseif W==11
    clear ny;

```

```

load be;

ny = [(be+0.1654)' zeros(1,3072) ];
elseif W==12
clear ny;
load bird;
ny = [(bi+0.1658)' zeros(1,7168) ];
elseif W==13
clear ny;
load boot;
ny = [(bo+0.1652)'];
elseif W==14
clear ny;
load call;
ny = [(cal+0.1654)' zeros(1,6144) ];
elseif W==15
clear ny;
load day;
ny = [(da+0.1645)' zeros(1,1024) ];
elseif W==16
clear ny;
load eka;
ny = [(ek+0.1653)'];
elseif W==17
clear ny;
load epa;
ny = [(ep+0.1650)' zeros(1,6144) ];
elseif W==18
clear ny;
load eve;
ny = [(ev+0.1654)' zeros(1,4096) ];
elseif W==19
clear ny;
load father;
ny = [(fa+0.1648)' zeros(1,6144) ];
elseif W==20
clear ny;
load foot;
ny = [(foo+0.1653)' zeros(1,6144) ];
elseif W==21
clear ny;
load for;
ny = [(fo+0.1649)' zeros(1,6144) ];
elseif W==22
clear ny;
load go;
ny = [(go+0.1651)'];
elseif W==23
clear ny;
load hate;
ny = [(ha+0.1657)' zeros(1,7168) ];
elseif W==24

clear ny;

```

```

load he
ny = [(he+0.1657)' zeros(1,2048) ];
elseif W==25
clear ny;
load ika
ny = [(ik+0.1654)' zeros(1,6144) ];
elseif W==26
clear ny;
load it

ny = [(it+0.1657)' zeros(1,3072) ];
elseif W==27
clear ny;
load key;
ny = [(ke + 0.1652)' zeros(1,2048)];
elseif W==28
clear ny;
load let;
ny = [(le + 0.1657)'];
elseif W==30
clear ny;
load met;
ny = [(met + 0.1653)'];
elseif W==31
clear ny;
load no;
ny = [(no + 0.1646)' zeros(1,1024)];
elseif W==34
clear ny;
load pay;
ny = [(pa + 0.1655)' zeros(1,1024)];

elseif W==36
load see;
ny = [(se+0.1653)' zeros(1,1024)];
elseif W==37
load she;
ny = [(sh+0.1654)'];
elseif W==38
load then;
ny = [(th+0.1656)' zeros(1,6144)];
elseif W==39
load thin;
ny = [(thi + 0.1655)' zeros(1,1024)];
elseif W==40
load to;
ny = [(to + 0.1649)' zeros(1,3072)];
elseif W==41
load up;
ny = [(up + 0.1653)' zeros(1,2048)];
elseif W==43
load vote;
ny = [(vo + 0.1654)' zeros(1,1024)];
elseif W==44

```

```

clear ny;
load we
ny = [(we+0.1655)' zeros(1,2048) ];
elseif W==45
clear ny;
load you
ny = [(you + 0.1655)' zeros(1,2048) ];
elseif W==46
clear ny;
load zoo
ny = [(zo+0.1646)' zeros(1,2048) ];
elseif W==47
clear ny;
load myvoice;
ny = x(1:8192)';
elseif W==48
clear ny;
load bye;
ny = [ bye' zeros(1,9216)];
elseif W==49
clear ny;
beback = loadwav('beback.wav');
ny = [ (beback/max(abs(beback))+0.056)' zeros(1,4824)];
elseif W==50
clear ny;
blows = loadwav('blows.wav');
ny = [ (blows(1:16384)/max(abs(blows))+0.0034)'];
elseif W==51
clear ny;

br = loadwav('bruna.wav');
ny = [ (br/max(abs(br)) + 0.0155)' zeros(1,7268) ];
elseif W==52
clear ny;
adam = loadwav('adamsfam.wav');
ny = [ (adam(1:32768)/max(abs(adam)) + 0.0081)'];

end
end
n = length(ny)
D = input('Enter the finest depth for Time Splitting : ');

% Implementing the Cosine Packet transform

cp = CPAnalysis(ny,D,'Sine');
stree = CalcStatTree(cp,'Entropy');
[btree,vtree] = BestBasis(stree,D); % Choosing the basis by applying the Best Basis Algorithm
[n,L] = size(cp);

% Create Bell

bellname = 'Sine';

```

```

    m = n / 2^D / 2;
    [bp,bm] = MakeONBell(bellname,m);
    x = zeros(1,n);

% initialize tree traversal stack
    stack = zeros(2,2^D+1);
    tp = zeros(1,n);
    v = zeros(1,n);
    compr = zeros(1,n);
    coef = zeros(1,n);
    ncoef = zeros(1,n);
    k = 1;
    stack(:,k) = [0 0]';
    v = zeros(1:n);
    ind = 0;
    le = zeros(1,2^D);
    while(k > 0),

        d = stack(1,k); b = stack(2,k); k=k-1;
        if(btrees(node(d,b)) ~= 0) , % nonterminal node
            k = k+1; stack(:,k) = [(d+1) (2*b)]';
            k = k+1; stack(:,k) = [(d+1) (2*b+1)]';
        else
            c = cp(packet(d,b,n),d+1)';
            coef(1,b/(2^d).*n+1:(b+1)/(2^d).*n) = c;
            i = (b/(2^d)*n+1);
            len = length(c);
            [ I,ND] = max(abs(c));
            compr(1,b/(2^d)*n+1) = length(c);

% Identifying the Frequency content

            if ND <= round(len/16)

                v(i)=0.4;

            elseif ND<= round(len/8)

                v(i) = 0.6;

            elseif ND <= round(len/(16/3))

                v(i) = 0.7;

            elseif ND <= length(c)/(2*P)

                v(i) = 0.8;

            else

                [sI,sND] =max(abs([coef(i:i+ND-3),0,0,coef(i+ND+1:i+len-1)]));
                if sND >= len/(2*P)
                    v(i) = 1;
                elseif sND > round(len/(16/3))

```



```

        v(i) = 0.8;
    elseif sND > round(len/8)
        v(i) = 0.7;
    elseif sND > round(len/16)
        v(i) = 0.6;
    else
        v(i) = 0.4;
    end

end

ec = sum(c.^2); % Computing the Energy of the Coefficients
vari = std(c);
tp (1,b/(2.^d).*n+1) = 1;
len = length(c);
ind = ind + 1;
le(ind) = log2(len);

v(i);
rko = length(c)/16;
ko = ND;
fo = 4000/length(c)*ND;

toten = sum(coef.^2);
i = (b/(2^d)*n+1);

% Applying the Adaptive Thresholding Compression Scheme

if v(i) <= 0.6
    if sum(coef(i:compr(i)-1+i).^2) < toten/n * len
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);
        else
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);
        end
    else
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);
        else
            nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),98.7);%98.7
        end
    end
end

nc = nncoef(i:compr(i)-1+i);

end

if v(i) > 0.6
    sumco = sum(coef(i:compr(i)-1+i).^2);
    thres = 0.5*toten/n * len;

```

```

if sum(coef(i:compr(i)-1+i).^2) < toten/n * len; % it was 0.5*toten/n*len
    if len < 2*n/(2^D)
        nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);%0.91% for all in "/be nice/"
    else
        nncoef(i:len+i-1) = comp(coef(i:len+i-1),99.5);
    end
else
    if len < 2*n/(2^D)
        nncoef(i:compr(i)-1+i) = comp((coef(i:compr(i)-1+i)),99.5);
    else
        nncoef(i:compr(i)-1+i) = comp(coef(i:compr(i)-1+i),99.5);
    end
end
end

```

```

nc = nncoef(i:compr(i)-1+i);

```

```

end

```

```

y = dct_iv(nc); % Inverse transforming each interval

```

% Unfolding each interval and Reconstructing the time sequence after compression

```

[xc,xl,xr] = unfold(y,bp,bm);
x(packet(d,b,n)) = x(packet(d,b,n)) + xc;
    if b>0,
        x(packet(d,b-1,n)) = x(packet(d,b-1,n)) + xl;
    else
        x(packet(d,0,n)) = x(packet(d,0,n)) + edgeunfold('left',xc,bp,bm);
    end
    if b < 2^d-1,
        x(packet(d,b+1,n)) = x(packet(d,b+1,n)) + xr;
    else
        x(packet(d,b,n)) = x(packet(d,b,n)) + edgeunfold('right',xc,bp,bm);
    end
end
end

```

```

end

```

```

nind = sum(le>0);

```

```

nle = le(1:nind);

```

```

figure(1),plot(ny) , hold;

```

```

plot(tp,''),hold off;

```

```

print figure(1)_deps

```

```

figure(2),plot(ny),hold

```

```

plot(v,''),hold off;

```

```

print -deps figure2

```

```

mse = mean((ny - x).^2) % computing the mean square error between the original and
                        % the reconstructed sequence;

```

```

scoefm0 = sum(abs(coef)>0) % computing the number of non-zero coefficients before
                        % compression

```

```

sncoefm0 = sum(abs(nncoef)>0) % computing the number of non-zero coefficients after
                        % compression

```

```

figure(3),
plot(x);

figure(4)
subplot(2,2,1),plot(ny,'b')
title('a)"ISSOS", ORIGINAL PLOT')
subplot(2,2,2),plot(x,'b')
title('c)AFTER FIXED THRESHOLDING(0.78%)')
subplot(2,2,3),specgram(ny,[],1)
title('b) SPECTOGRAM')
subplot(2,2,4),specgram(x,[],1)
title('d) SPECTOGRAM (AFTER)')

```

```

% Name: encp6.m and ndencomp.m
% Subject: Analysis, Denoising/Compression and Synthesis of Speech data;
% Description: These two routines contain the Denoising scheme applied prior to
% the compression schemes;
% The differences between the two routines are in:
% a) The Frequency Identification implementation; for example ndencomp.m
%    makes more use of the second largest coefficient than encp6.m does;
% b) The segmentation between voiced and unvoiced segments: encp6.m uses
%    500 Hz for female speech and 1,000 Hz for male speech; ndencomp.m
%    uses 1,000Hz for any gender;
%
% c) The detection of the presence of low energy speech in high
%    energy noisy background. ndencomp.m implements such a scheme, while encp6.m
%    doesn't;
%
% d) The Adaptive Thresholding Compression Scheme
% Written and adapted by J. Roberto V. Martins, October 1995;

```

```

% Encp6.m

```

```

clear;

```

```

% Input and loading of speech data

```

```

V = input('Please enter "1" for female voice and "2" for a male voice : ');
if V==1
    P = 8;
    FV = input('Please enter 1 for the sentence, 2 for "be" , 3 for "hate", 4 for "hey" , 5 for "met" , 6 for
"pay", 7 for "cats", 8 for "benice" : ');
    if FV == 1

```

```

clear ny;
load fse;
ny = [fse' zeros(1,5120)];

elseif FV==2
clear ny;
load fbe;
ny = [fbe' zeros(1,2048)];
elseif FV==3
clear ny;
load fha;
ny = [fha' zeros(1,7168)];
elseif FV==4
clear ny;
load fthey;
ny = [fthey'];
elseif FV==5
clear ny;
load fmet;
ny = [fmet' zeros(1,1024)];
elseif FV==6
clear ny;
load fpay;
ny = [fpay' zeros(1,1024)];

elseif FV==7
clear ny;
load fcats;
ny = [ fcats'];
elseif FV==8
clear ny;
benice = loadwav('benice.wav');
ny = [((benice(1:16384)/max(benice)) + 0.0119)'];
end
end
if V==2
P = 4;
W = input('Please enter 1,for "project",2 for "cataratas",3 for "encyclopedia", 4 for "issos",5 for "assos",6
for "six",7 for "the sentence",8 for "aka",9 for "at",10 for "azure",11 for "be",12 for "bird",13 for "boot",14
for "call",15 for "day",16 for "eka",17 for "epa", 18 for "eve",19 for "father",20 for "foot", 21 for "for", 22
for "go", 23 for "hate", 24 for "he",25 for "ika",26 for "it",27 for "key",28 for "let",29 for "me",30 for
"met",31 for "no",32 for "obey",33 for "opa",34 for "pay",35 for "read",36 for "see",37 for "she",38 for
"then",39 for "thin", 40 for "to",41 for "up", 43 for "vote",44 for "we", 45 for "you", 46 for "zoo",47 for
"silence", 48 for "the bye sentence",49 for "beback", 50 for "blows", 51 for "bruna", 53 for "sounds good" :
');

if W==1
clear ny;
load newvoice;
ny = y(2700:2700+8191);
elseif W==2
clear ny;
load catar;

```

```

    ny = ca(1900:1900+8191)';
elseif W==3
    clear ny;
    load encic;
    ny = en(1200:1200 +8191)';
elseif W==4
    clear ny;
    load issos
    ny = is(1900:1900+8191)';
elseif W==5
    clear ny
    load assos
    ny = as(1900:1900+8191)';
elseif W==6
    clear ny
    load six
    ny = si(1:8192)';
elseif W==7
    clear ny;
    load myvoice;
    ny = x(9000:9000+32767)';
elseif W==8
    clear ny;
    load aka;
    ny = (ac+0.1656)';
elseif W==9
    clear ny;
    load at;
    ny = (at+0.1655)';

elseif W==10
    clear ny;
    load azure;
    ny = [(az+0.1651)' zeros(1,6144) ];
elseif W==11
    clear ny;
    load be;
    ny = [(be+0.1654)' zeros(1,3072) ];
elseif W==12
    clear ny;
    load bird;
    ny = [(bi+0.1658)' zeros(1,7168) ];
elseif W==13
    clear ny;
    load boot;
    ny = [(bo+0.1652)'];
elseif W==14
    clear ny;
    load call;
    ny = [(cal+0.1654)' zeros(1,6144) ];
elseif W==15
    clear ny;
    load day;
    ny = [(da+0.1645)' zeros(1,1024) ];

```

```

elseif W==16
    clear ny;
    load eka;
    ny = [(ek+0.1653)'];
elseif W==17
    clear ny;
    load epa;
    ny = [(ep+0.1650)' zeros(1,6144) ];
elseif W==18
    clear ny;
    load eve;
    ny = [(ev+0.1654)' zeros(1,4096) ];
elseif W==19
    clear ny;
    load father;
    ny = [(fa+0.1648)' zeros(1,6144) ];
elseif W==20
    clear ny;
    load foot;
    ny = [(foo+0.1653)' zeros(1,6144) ];
elseif W==21
    clear ny;
    load for;
    ny = [(fo+0.1649)' zeros(1,6144) ];

elseif W==22
    clear ny;
    load go;
    ny = [(go+0.1651)'];
elseif W==23
    clear ny;
    load hate;
    ny = [(ha+0.1657)' zeros(1,7168) ];
elseif W==24
    clear ny;
    load he
    ny = [(he+0.1657)' zeros(1,2048) ];
elseif W==25
    clear ny;
    load ika
    ny = [(ik+0.1654)' zeros(1,6144) ];
elseif W==26
    clear ny;
    load it
    ny = [(it+0.1657)' zeros(1,3072) ];
elseif W==27
    clear ny;
    load key;
    ny = [(ke + 0.1652)' zeros(1,2048)];
elseif W==28
    clear ny;
    load let;
    ny = [(le + 0.1657)'];
elseif W==30

```



```

clear ny;
load met;
ny = [(met + 0.1653)'];
elseif W==31
clear ny;
load no;
ny = [(no + 0.1646)' zeros(1,1024)];
elseif W==34
clear ny;
load pay;
ny = [(pa + 0.1655)' zeros(1,1024)];
elseif W==36
load see;
ny = [(se+0.1653)' zeros(1,1024)];
elseif W==37
load she;
ny = [(sh+0.1654)'];
elseif W==38
load then;
ny = [(th+0.1656)' zeros(1,6144)];
elseif W==39
load thin;
ny = [(thi + 0.1655)' zeros(1,1024)];
elseif W==40
load to;
ny = [(to + 0.1649)' zeros(1,3072)];
elseif W==41
load up;
ny = [(up + 0.1653)' zeros(1,2048)];
elseif W==43
load vote;
ny = [(vo + 0.1654)' zeros(1,1024)];
elseif W==44
clear ny;
load we
ny = [(we+0.1655)' zeros(1,2048) ];
elseif W==45
clear ny;
load you
ny = [(you + 0.1655)' zeros(1,2048) ];
elseif W==46
clear ny;
load zoo
ny = [(zo+0.1646)' zeros(1,2048) ];
elseif W ==47
clear ny;
load myvoice;
ny = x(1:8192)';
elseif W ==48
clear ny;
load bye;
ny = [ bye' zeros(1,9216)];
elseif W ==49
clear ny;

```

```

load beback;
ny = [ (beback-127.4452)' zeros(1,4824)];
elseif W ==50
clear ny;
blows = loadwav ('blows.wav');
ny = [ (blows(1:16384)+0.3027)'];
elseif W ==51
clear ny;
br = loadwav('bruna.wav');
ny = [ (br/max(abs(br))+0.0155)' zeros(1,7268)];
elseif W ==52
clear ny;
br = loadwav('adamsfam.wav');
ny = [ (adam(1:32768)/max(abs(adam))+0.0081)'];
elseif W ==53
clear ny;
load engl6;
ny = [ (engl6(1:16384) + 3.019e-4)'];
elseif W ==54
clear ny;
load voiq;
ny = [ voiq(1:32768)'];
end
end

```

% Implementing the Cosine Packet Transform

```

n = length(ny)
D = input('Enter the finest depth for Time Splitting : ');
cp = CPAnalysis(ny,D,'Sine');
stree = CalcStatTree(cp,'Entropy');
[btree,vtree] = BestBasis(stree,D);
[n,L] = size(cp);

```

% Create Bell

```

bellname = 'Sine';

m = n / 2^D /2;
[bp,bm] = MakeONBell(bellname,m);
%
x = zeros(1,n);
%
% initialize tree traversal stack
%
stack = zeros(2,2^D+1);
tp = zeros(1,n);
v = zeros(1,n);
compr = zeros(1,n);
coef = zeros(1,n);
ncoef = zeros(1,n);
k = 1;

```

```

    stack(:,k) = [0 0]';
    v = zeros(1:n);
    ind = 0;
    le = zeros(1,2^D);
    while(k > 0),

        d = stack(1,k); b = stack(2,k); k=k-1;
        if(btree(node(d,b)) ~= 0) , % nonterminal node
            k = k+1; stack(:,k) = [(d+1) (2*b) ]';
            k = k+1; stack(:,k) = [(d+1) (2*b+1)]';
        else
            c = cp(packet(d,b,n),d+1)';
            coef(1,b/(2^d).*n+1:(b+1)/(2^d).*n) = c;
            i = (b/(2^d).*n+1);
            len = length(c);
            [ I,ND] = max(abs(c));
            compr(1,b/(2^d).*n+1) = length(c);

% Identifying the Frequency Content of each interval

            if ND <= round(len/16)
                [sI,sND] = max(abs([coef(i:i+ND-2),0,coef(i+ND:i+len-1)]));

                if (4000/len*sND) > 125 %ND <= round(len/32)

                    if (4000/len*sND) < 400

                        if (4000/len*sND) >= 125
                            v(i) = 1;
                            ncoef(i:compr(i)-1+i) =
[zeros(1,round(len/64)),coef(i+round(len/64):i+round(len/5)-1),zeros(1,len-round(len/5)) ]; % Denoising
                        else

                            if (4000/len*sND) >= 60
                                ncoef(i:compr(i)-1+i) =
[zeros(1,round(len/64)),coef(i+round(len/64):i+round(len/16)-1),zeros(1,len-round(len/16)) ];%Denoising
                                v(i) = 1;
                            else
                                if (4000/len*sND) <=30
                                    ncoef(i:len-1+i) = zeros(1,len); % Denoising
                                    v(i) = 0;
                                else
                                    ncoef(i:len-1+i) = [zeros(1,round(len/64)), coef(i+round(len/64):
i+round(len/16)-1),zeros(1,len-round(len/16)) ]; % zeros(1,len);
                                    v(i) = 1;
                                end
                            end
                        end
                    end
                end
            elseif sND < len/4

```

```

ncoef(i:compr(i)-1+i)=[zeros(1,len/64),coef(i+round(len/64):i+len-1)];%Denoise
v(i) = 1;
else

    ncoef(i:compr(i)-1+i) = [ zeros(1,len/16),coef(i+len/16:i+len-1) ]; % Denoising
    v(i) = 2;
end
end
if (4000/len*ND) <= 125
    if sND <= len/16
        ncoef(i:len-1+i) = zeros(1,len); % Denoising
        v(i) = 0;
    elseif sND <= len/4
        if (4000/len*sND) >= 300
            ncoef(i:i+len-1) = [zeros(1,sND-1),coef(i+sND-1),zeros(1,len-
sND)];%[zeros(1,len/16),coef(i+len/16:i+len-1) ];
            v(i) = 1;
        else
            ncoef(i:i+len-1) = zeros(1,len); % Denoising
            v(i) = 0;
        end
    else
        if (4000/len*ND) < 64
            ncoef(i:compr(i)-1+i) = zeros(1,len); % Denoising
            v(i)=0;
        else
            ncoef(i:compr(i)-1+i) = [ zeros(1,len/4),coef(i+len/4:i+len-1) ];
            v(i) = 2;
        end
    end
end
end

elseif ND < length(c)/P
    [sI,sND] = max(abs([coef(i:i+ND-2),0,coef(i+ND:i+len-1)]));
    if sND < len/32
        SND =sND
        ncoef(i:compr(i) - 1+i) = zeros(1,len); % Denoising
        v(i) = 0;
    else
        v(i) = 1;
        ncoef(i:compr(i)-1+i)=[ zeros(1,len/16),coef(i+len/16:i+len-1) ]; % Denoising
    end
else
    [sI,sND] = max(abs([coef(i:i+ND-3),0,0,0,coef(i+ND+1:i+len-1)]));
    if sND >= length(c)/(2*P)
        v(i) = 2;
        ncoef(i:compr(i)-1+i) = [ zeros(1,len/16),coef(i+len/16:i+len-1) ]; % Denoising
    else
        v(i) = 1;
        ncoef(i:compr(i)-1+i) = [ zeros(1,len/16),coef(i+len/16:i+len-1) ]; % Denoising
    end
end
end
ec = sum(c.^2);
vari = std(c);

```

```

tp (1,b/(2.^d).*n+1) = 1;
len = length(c);
ind = ind + 1;
le(ind) = log2(len);
v(i);
rko = length(c)/16;
ko = ND;
fo = 4000/length(c)*ND;

```

```

de(ind) = d;
be(ind) = b;
toten = sum(coef.^2);
i = (b/(2.^d)*n+1);
if v(i) == 0
    nncoef(i:compr(i)-1+i) = ncoef(i:compr(i)-1+i);
    nc = nncoef(i:compr(i)-1+i);
end

```

% Applying the Adaptive Thresholding Compression Scheme

```

if v(i) == 1

```

```

    if sum(ncoef(i:compr(i)-1+i).^2) < toten/n * len
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);
        else
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);
        end
    else
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),98.7);
        else
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),97.66);
        end
    end
end

```

```

    nc = nncoef(i:compr(i)-1+i);

```

```

end

```

```

if v(i) == 2
    sumco = sum(coef(i:compr(i)-1+i).^2);
    thres = 0.5*toten/n * len;
    if sum(coef(i:compr(i)-1+i).^2) < toten/n * len; % it was 0.5*toten/n*len
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);
        else
            nncoef(i:len+i-1) = comp(ncoef(i:len+i-1),99.5);
        end
    else
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);

```

```

        else
            nncoef(i:compr(i)-1+i) = comp(ncoef(i:compr(i)-1+i),99.5);
        end
    end

    nc = nncoef(i:compr(i)-1+i);

end

y = dct_iv(nc); % Inverse Transforming each interval

% Unfolding and reconstructing the time sequence after compression

    [xc,xl,xr] = unfold(y,bp,bm);
    x(packet(d,b,n)) = x(packet(d,b,n)) + xc;
    if b>0,
        x(packet(d,b-1,n)) = x(packet(d,b-1,n)) + xl;
    else
        x(packet(d,0,n)) = x(packet(d,0,n)) + edgeunfold('left',xc,bp,bm);
    end
    if b < 2^d-1,
        x(packet(d,b+1,n)) = x(packet(d,b+1,n)) + xr;
    else
        x(packet(d,b,n)) = x(packet(d,b,n)) + edgeunfold('right',xc,bp,bm);
    end
end

end

nind = sum(le>0);
nle = le(1:nind);
XX = x.*6;
figure(1),plot(ny) , hold;
plot(tp,':'),hold off;
figure(2),plot(ny),hold
plot(v,':'),hold off;
mse = mean((ny - x).^2) % Computing the mean square error between the
                        % original and the reconstructed compressed one
scoefm0 = sum(abs(coef)>0)
% Computing the number of non-zero
                        % coefficients before denoising/compression
sncoefm0 = sum(abs(nncoef)>0) % Computing the number of non-zero
                        % coefficients after denoising/compression

figure(3),
subplot(2,2,1),plot(ny,'b');
title('MET, male speaker');
subplot(2,2,2),plot(x,'b');
title('AFTER DENOISING/COMPRESSION')
subplot(2,2,3),specgram(ny);
title('Original Spectrogram');
subplot(2,2,4),specgram(x);
title('AFTER DENOISING/COMPRESSION')

```



```

% Ndencomp.m
% Obs.: This routine has parts a), b) and c) identical to the same parts of
% routine encp6.m. Thus we are only presenting the complement, which begins
% in part d).

complex = 0;
endearly = 0;
% Identifying the Frequency Content of each interval

if ND <= round(len/16); %it was len/64*3

    if ND <= round(len/64)
        [ sI,sND ] = max(abs([coef(i:i+ND-2),0,coef(i+ND:i+len-1)]));
        if (4000/len* sND) <= 300 % try to make it better!!!
            if len > n/(2^D)*8

                coef(i+ND-1)=0;
                coef(i+sND-1)=0;
                [ tI,tND ] = max(abs(coef(i:i+len-1)));% (recovering the "ts" sound)
                % implemented to solve problems like in "cats" : it
                % still needs to be improved!!

                if tND < round(len/20)

                    v(i) = 0.1;

                else
                    TND = tND;
                    complex = 1;
                    endearly = 1;
                    ncoef(i:i+len-1)=zeros(1,len);%[zeros(1,len/64),coef(i+len/64:i+len-1)];% Denoising

                    %[zeros(1,tND-1),coef(i+tND-1),zeros(1,len-tND)];%[zeros(1,len/64),coef(i+len/64:i+len-
                    1)];%[zeros(1,tND-1),coef(i+tND-1),zeros(1,len-tND)];%[zeros(1,tND-1),coef(i+tND-1:i+len-1)];

                    v(i) = 0.5;
                end

            else
                v(i) = 0.1;
            end
        elseif sND <= round(len/8)
            ncoef(i:i+len-1) = [ zeros(1,sND-1),coef(i+sND-1),zeros(1,len-sND) ];% Denoising
            v(i) = 0.5;
        elseif sND <= round(len/4)
            ncoef(i:i+len-1) = [ zeros(1,sND-1),coef(i+sND-1),zeros(1,len-sND) ];% Denoising
            v(i) = 1.0;
        else

            v(i) = 0.1;
        end

    else

        v(i) = 0.1;
    end

end

```

```

[ sI,sND ] = max(abs([coef(i:i+ND-2),0,coef(i+ND:i+len-1)]));
if sND < len/20    % see in encp6 how it was made for 125<ND<=250 & 125<=sND<400
    v(i) = 0.1;

elseif sND<= round(len/8)
    compless = 1; % flag to indicate to compress less
    ncoef(i:i+len-1) = [zeros(1,len/32), coef(i+len/32:i+len-1)];% Denoising
    v(i) = 0.5;
elseif sND < length(c)/(2*P)
    compless = 1;
    ncoef(i:i+len-1) = [zeros(1,len/32),coef(i+len/32:i+len-1)] ; % Denoising
    v(i) = 1;
elseif sND<= round(len/P)
    compless = 1;
    ncoef(i:i+len-1) = [zeros(1,len/32),coef(i+len/32:i+len-1)] ; % Denoising
    v(i) = 1.5;

else
    compless = 1;
    ncoef(i:i+len-1) = [zeros(1,len/32),coef(i+len/32:i+len-1)]; % Denoising
    v(i) = 2;

end
end
elseif ND <= round(len/8)
    ncoef (i:i+len-1)= [zeros(1,len/32) , coef(i+len/32:i+len-1)];% Denoising
    v(i) = 0.25; % it was 0.2
elseif ND<= round(len/4)
    ncoef(i:i+len-1) = [ zeros(1,len/32) , coef(i+len/32:i+len-1)] ;% Denoising
    v(i) = 0.5; % it was 0.4

elseif ND < length(c)/(2)
    ncoef(i:i+len-1) = [ zeros(1,len/32), coef(i+len/32:i+len-1)]; % Denoising
    v(i) = 1; % it was 0.6
else
    [sI,sND] = max(abs([coef(i:i+ND-3),0,0,0,coef(i+ND+1:i+len-1)]));
    if sND >= length(c)/(2*P)
        if ND <= len/2
            ncoef(i:i+len-1) = [ zeros(1,len/32), coef(i+len/32:i+len-1)] ;% Denoising
            v(i) = 1.5; %it was 0.75;
        elseif ND <= round(len*3/4)
            %compless = 1; % included to help in the voice quality sentence
            ncoef(i:i+len-1) = [zeros(1,len/32),coef(i+len/32:i+len-1)];% Denoising
            v(i) = 2; % it was 0.9
        else
            ncoef(i:i+len-1) = [zeros(1,len/32),coef(i+len/32:i+len-1)];% Denoising
            v(i) = 2.5; % it was 1.0
        end
    elseif sND > round(len/8)
        ncoef(i:i+len-1) = [zeros(1,len/32),coef(i+len/32:i+len-1)];% Denoising
        v(i) = 1; % it was 0.6

    elseif sND >= round(len/16)
        ncoef(i:i+len-1) = [zeros(1,len/32),coef(i+len/32:i+len-1)];% Denoising

```

```

        v(i) = 0.5; % it was 0.4
    else
        ncoef(i:i+len-1)=[ zeros(1,len/32),coef(i+len/32:i+len-1)];% Denoising
        v(i) = 0.25; % it was 0.2
    end

end

EC = sum(c.^2);
% Computing the coefficients energy
ec(i:i+len-1) = ones(1,len) .* sum(c.^2);
es(i:i+len-1) = ones(1,len) .* sum(ny(i:i+len-1).^2);
% Computing the energy of each interval

vari = std(c);
%nccoef(1,b/(2^d).*n+1:(b+1)/(2^d).*n) = nc;
tp (1,b/(2.^d).*n+1) = 1;
len = length(c);
ind = ind +1;
le(ind) = log2(len);
rko = length(c)/16;
ko = ND;
fo = 4000/length(c)*ND
de(ind)=d;
be(ind)=b;
toten = sum(coef.^2);
i = (b/(2^d).*n+1);

% Applying the Adaptive Thresholding Compression Scheme

if v(i) == 0.1

    nncoef(i:i+len-1) = zeros(i:i+len-1);
    nc = nncoef(i:compr(i)-1+i);
elseif v(i) <= 0.5
    if sum(coef(i:compr(i)-1+i).^2) < toten/n * len
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);
        else
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);
        end
    else
        if len < 2*n/(2^D)
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),98.7);
        else
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),97.66);
        end
    end
end

nc = nncoef(i:compr(i)-1+i);

end

if v(i) > 0.5 % it was 0.4; % it was == 1

```

```

sumco = sum(coef(i:compr(i)-1+i).^2);
thres = 0.5*toten/n * len;
if sum(coef(i:compr(i)-1+i).^2) < toten/n * len; % it was 0.5*toten/n*len
    if len < 2*n/(2^D)
        if compleness == 1
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),97.66);
        else
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);
        end
    else
        if compleness == 1
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),98.7);
        else
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);
        end

    end
else
    if len < 2*n/(2^D)
        if compleness == 1
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),98.7);
        else
            nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),99.5);
        end
    else
        %if compleness == 1
        %nncoef(i:compr(i)-1+i) = comp((ncoef(i:compr(i)-1+i)),98.7);
        %else
        nncoef(i:compr(i)-1+i) = comp(ncoef(i:compr(i)-1+i),99.5);
        %end
    end
end

nc = nncoef(i:compr(i)-1+i);

end
if endearly == 1
    [ma,md] = max(nncoef(i:i+len-1));
end
y = dct_iv(nc);
% Inverse Transforming each interval
if endearly == 1
    y = y.*(abs(nncoef(i:i+len-1))>0);
end
% Unfolding and Reconstructing the Time sequence after compression
[xc,xl,xr] = unfold(y,bp,bm);
x(packet(d,b,n)) = x(packet(d,b,n)) + xc;
    if b>0,
        x(packet(d,b-1,n)) = x(packet(d,b-1,n)) + xl;
    else
        x(packet(d,0,n)) = x(packet(d,0,n)) + edgeunfold('left',xc,bp,bm);
    end
    if b < 2^d-1,

```

```

        x(packet(d,b+1,n)) = x(packet(d,b+1,n)) + xr;
    else
        x(packet(d,b,n)) = x(packet(d,b,n)) + edgeunfold('right',xc,bp,bm);
    end
end

end

nind = sum(le>0);
nle = le(1:nind);
figure(1),plot(ny) , hold;
plot(tp,':'),hold off;
figure(2),plot(ny),hold
plot(v,':'),hold off;
mse = mean((ny - x).^2)

% Computing the mean square error between original signal and the signal after compression

scoefm0 = sum(abs(coef)>0)
% Computing the number of non-zero-coefficients before
% denoising/compression
sncoefm0 = sum(abs(nncoef)>0)
% Computing the number of non-zero coefficients after
% denoising/compression

figure(3),
plot(x);
figure(4),
plot(ec);
figure(5),
plot(es);
first = 1;
nv = zeros(1,length(v));
nnv = zeros(1,length(v));
for i=1:length(v)
    if v(i) > 0
        if v(i) <= 0.5
            nv(i) = 1.0;
            dist = i - first;
            %if dist >= 512
            if ec(first) > toten/(n*32)*dist
                nnv(first) = nv(first);
                nnv(i) = nv(i);
            end
            first = i;
        end
        if v(i) > 0.5
            nnv(i) = 1.5;
        end
    end
end
XX = x.*4;
figure(6),plot(ny),hold
plot(nnv,':'),hold off;
figure(7)
subplot(2,2,1),plot(ny,'b')

```

```

title('PAY, male speaker')
subplot(2,2,2),plot(x,'b')
title('AFTER DENOISING/COMPRESSION ')
subplot(2,2,3),specgram(ny,[],1)
title('ORIGINAL SPECTOGRAM')
subplot(2,2,4),specgram(x,[],1)
title('AFTER DENOISING/COMPRESSION')

```

```

% Name: encptour.m and ndentour.m
% Subject: Analysis, Denoising/Compression, Encoding, Decoding and Synthesis
% of speech data
% Description: These two routines were applied on top of encp6.m and
% ndencomp.m. These two programs perform the following tasks in addition to those
% already performed by encp6.m and ndencomp.m:

```

```

% a) Implementation of the Linear Quantizer for the Coefficients
% vector;

```

```

% b) Encoding of the Locations Vector;

```

```

% c) Encoding of the positions of begining of each segment;

```

```

% d) Huffman coding of Coefficients Vector and for Locations vector;

```

```

% e) Decoding of all the vectors on the Receiver's side

```

```

% f) Reconstruction of the Denoised/compressed sequence at the
% receiver's side;

```

```

% Obs.: That code is put on top of the existent codes encp6.m and ndencomp.m
% Written by J. Roberto V. Martins, October 1995.

```

```

[X,L,seglens,de,be] = enc(nncoef,nle,de,be); % Encoding the locations and coefficients

```

```

[TX,prob,nprob,probdesc,N,nq,S] = quantx(X,QL); % Quantizing the coefficients

```

```

np = length(probdesc);

```

```

avwcoeff = huffcod(np,probdesc); % Huffman coding the coefficients vector

```

```

totcoeff = avwcoeff*length(TX);

```

```

debe = [ de be ];

```

```

sdebe = size(debe);

```

```

ordebe = sort(debe);

```

```

ndebe = zeros(1,length(ordebe));

```

```

countdb = 1;

```

```

ndebe(1,1) = ordebe(1);

```

```

lndb = length(ndebe);

```

```

for countdb=1:length(debe)-1

```



```

if ordebe(countdb+1) > ordebe(countdb)

    countdb=countdb+1;
    ndebe(countdb) = ordebe(countdb);

end
end
index = 0;
czero = 0;
for countdb =1:length(ndebe)

    if ndebe(countdb) > 0

        index = index +1;

        nndebe(index) = ndebe(countdb);

    else

        czero = czero + 1;

    end

end

end
probdebe = nndebe/sum(nndebe);

probdbde = fliplr(sort(probdebe));

nprobdbd = probdbde(1:length(probdbde));

avwdebe = huffcod(length(nndebe),nprobdbd); % coding the des and the bes (see chapter VII)
totdebe = avwdebe*(length(debe) - czero) + czero;

[DL,probl,lenprob] = difl(L);

% mDL = max(DL(2:length(DL)));

totndl = 0;

pow = 1;

for indl =1:length(DL)

    while DL(indl) > 2^pow,

        pow = pow+1;

    end

    totndl = totndl + pow; % calculating the necessary number of bits to transmit
    NDLwe're not using this
    pow = 1;
end

```

```
totmdl = totmdl + round(log2(DL(1))) ;% we're using the vector DL to transmit the
locations
```

```
sn = 0;
```

```
qc = zeros(1,n);
```

```
nde = fliplr(de);
```

```
nbe = fliplr(be);
```

```
nseglens = fliplr(seglens);
```

```
nv = 1;
```

```
I = nbe./(2.^nde).*n + 1;
```

```
for ns = 1:length(L)
```

```
    for ni = 1:length(I)-1
```

```
        if L(ns) >= I(ni)
```

```
            if L(ns) <= I(ni+1)
```

```
                sn = sn+1;
```

```
                SN(sn) = ni;
```

```
                NDL(sn) = L(ns) - I(ni);
```

```
            end
```

```
        end
```

```
    end
```

```
end
```

```
pro = SN/sum(SN);
```

```
nsimbsn = max(SN) - min(SN) + 1;
```

```
realpr = zeros(1,nsimbsn);
```

```
indsn = 1;
```

```
realpr(indsn) = pro(indsn);
```

```

for isn = 1:length(pro)-1

    if pro(isn+1) == pro(isn)

        realpr(indsn) = realpr(indsn) + pro(isn+1);

    else

        realpr(indsn+1) = pro(isn+1);

        indsn = indsn + 1;

    end
end
desreapr = fliplr(sort(realpr));

RL(1) = DL(1);    % Reconstructing L, the locations vector

for cl = 1:length(DL)-1

    RL(cl+1) = RL(cl) + DL(cl+1);

end

for nv=1:length(nde)-1                %i:i+(b/(2^d)*n-1)  1:length(nseglens)

    d = nde(nv);

    b = nbe(nv);

    i = (b/(2^d)*n+1);

    nnc = qc(i:(nbe(1,nv+1)/(2^nde(1,nv+1))*n));    %(i:2^nseglens(v)+i-1);

    thislen = nbe(nv+1)/2^nde(nv+1)*n-i+1;

    for z = i:i + (thislen-1)                %(nbe(1,nv+1)/(2^nde(1,nv+1))*n) %1:2^nseglens(nv)

        for t = 1:length(RL)

            if z==RL(t)

                qc(z) = TX(t)/(nq/2)*S;

            end

        end

    end
end

```

```

unfprev = 0;

unfnex = 0;

nnc = qc(i:(nbe(1,nv+1)/(2^nde(1,nv+1))*n)); % (i:2^nseglens(v)+i-1);

% Inverse transforming to the Time Domain, Unfolding and Reconstructing the Denoised/Compressed
Decoded Speech Sequence

y = dct_iv(nnc);
[xc,xl,xr] = unfold(y,bp,bm);

xl(packet(d,b,n)) = xl(packet(d,b,n)) + xc;
    if nv > 1          %nv == 1      %if b>0,

        xl(packet(d,b-1,n)) = xl(packet(d,b-1,n)) + xl;
    else
        xl(packet(d,b,n)) = xl(packet(d,b,n)) + edgeunfold('left',xc,bp,bm);
    end
    if b < 2^d-1,

        xl(packet(d,b+1,n)) = xl(packet(d,b+1,n)) + xr;
    else

        xl(packet(d,b,n)) = xl(packet(d,b,n)) + edgeunfold('right',xc,bp,bm);

    end

end

figure(5),plot(ny),hold

plot(v,':'),hold off;

mse = mean((x - x1).^2) % Computing the mean square error between the denoised/compressed in
                        % the trasmitter and the decoded sequence in the receiver;
scoefm0 = sum(abs(coef)>0) % Computing the number of original non-zero coefficients
sncoefm0 = sum(abs(nncoef)>0) % Computing the number of non-zero coefficients
                        % afterdenoising/compression
sqcoefm0 = sum(abs(qc)>0) % Computing the number of non-zero coefficients after decoding
figure(6),
plot(x);
figure(7)

subplot(2,3,1),plot(ny)

title('Original "PAY", Female speaker')
subplot(2,3,2),plot(x)
title('After Denoising/Compression')
subplot(2,3,3),plot(x1)

title('After Decoding')

subplot(2,3,4),specgram(ny,[],1)

```

```

title('Original Spectrogram')
subplot(2,3,5),specgram(x,[],1)
title('After Denoising/Compression')
subplot(2,3,6),specgram(x1,[],1)

```

```

title('After Decoding')

```

```

TOTNBITS = totcoeff + totdebe + totndl

```

```

TOTNSAMP = length(TX) + length(debe) + length(SN) + length(NDL)

```

```

BITPSAMP = TOTNBITS/TOTNSAMP

```

```

COMPRATIO = 100 - (TOTNBITS/(scoefm0*8)*100)

```

```

% Name: Comp.m
% This function receives as input:
% A vector "c" composed of coefficients and
% a percentage number "pcent";
% As an output, this function gives a vector of same length which the non-zero
% components are the top % dominant (100 - pcent) pcent coefficients extracted from
% that original vector;
% Written by J. Roberto V. Martins in October of 1995.

```

```

function cc = comp(c,pcent)

```

```

d = sort(abs(c));
p = round(pcent/100*length(c));
for i = 1:length(c)
if p==0
cc(i) = c(i);
elseif abs(c(i)) <= d(p)
cc(i) = 0;
else cc(i) = c(i);
end
end
%d = (abs(c)>pcent/100*max(abs(c)));
%cc = c.*d;

```

```

% Name: Enc.m
% This function receives a vector, its length and the vectors de
% and be. As an output, it returns:
% X, a vector with non-zero coefficients extracted from the input vector;
% L, the vector containing the locations of the non-zero coefficients from
% the original input vector;
% Written by J. Roberto V. Martins in October of 1995

```

```

function [X,L,seglens,de,be] = enc(vector,lenvec,de,be)

n = 0;
m=0;

for i = 1:length(vector)
    if abs(vector(i)) > 0
        n = n + 1;
        X(n) = vector(i);
    end
end

for j = 1:length(vector)
    if abs(vector(j)) > 0
        m = m + 1;
        L(m) = j;
    end
end

seglens = lenvec;

% Name: Difl.m
% This function encodes a vector by transforming it into a
% differentially encoded vector. It receives the vector to be encoded as an input and returns :
% _The differences vector
% _The probabilities vector in descending order as well as its length

function [DL,prob,lendl] = difl(vec)

DL(1) = vec(1);

for z = 2:length(vec)

    DL(z) = vec(z) - vec(z-1);

end
a=0;
N = zeros(1,length(DL));

count = 1;
SDL = sort(DL);
NSDL(1) = SDL(1);
for p = 1:length(SDL)-1
    if SDL(p+1) > SDL(p)
        count = count + 1;
        NSDL(1,count) = SDL(1,p+1);
    end
end
N = zeros(1,length(NSDL));
for l = 1:length(DL)

```



```

for x = 1:length(NSDL)

    if DL(l)==NSDL(x)

        N(x) = N(x) + 1;

    end

end

end

end

prob = fliplr(sort(N)/sum(N));

lendl = length(prob);

% Name: Quantx.m
% This function performs the Linear Quantization proposed in this thesis for a
% given input vector. Inputs are the vector X to be quantized and
% the number of quantization levels desired, nq.
% Outputs are:
% _ TX: the quantized vector to be transmitted;
% _ prob: The vector of probabilities of all values in the input vector;
% _ nprob: The new vector of probabilities of all non-zero values in the input vector;
% _ probdesc: The new probabilities vector in descending order for input to Huffman code;
% _ N: The length of probdesc;
% _ nq: The number of quantization levels (equal to the input nq);
% _ S: The scaling factor S, i.e. the highest present absolute value in the vector;

% Written by J.Roberto V. Martins, October 1995.

function [TX,prob,nprob,probdesc,N,nq,S] = quantx(X,nq)

prob = zeros(1,nq+1);

S= max(abs(X));
normX = X/S;

TX = round(normX*nq/2);

%[N,Q] = hist(TX,length(TX));

```

```

N = zeros(1,nq+1);

STX = sort(TX);

for s = 1:length(TX)

    for p = -nq/2:1:nq/2

        if TX(s) == p
            N(1,p+nq/2+1) = N(1,p+nq/2+1) + 1;
        end

    end

end

prob = N/sum(N);
t = 0;
for s=1:length(prob)

    if prob(s) > 0

        t = t+1;
        nprob(1,t)=prob(1,s);

    end

end

probdesc = fliplr(sort(nprob));

```

```

% Name: Huffcod.m
% This function receives as input :
% q , the number of symbols; and
% p , the vector containing the probabilities of each symbol;
% As an output, it gives the average word length of the sequence;
% The function uses the code Huffman.m, by K.L. Frack written on 30 November 1993
% Modifications made by J.Roberto V. Martins in October 1995.

```

```

% HUFFMAN finds the minimum variance Huffman code for the symbol
% probabilities entered by the user. The algorithm makes use of
% permutation matrices for the combination and sorting of probabilities.
% Permutation matrices are used because they provide a convenient record
% of operations, so that the codewords can then be constructed fairly easily
% once the combination and sorting of probabilities yields just two
% probabilities. At this point a zero is assigned to one of the
% probabilities and a one assigned to the other. The permutation matrices
% are used to append additional zeros and ones as appropriate to obtain
% the final codeword for each symbol.

```

%% Written by K.L. Frack for EC4580 Course Project

% Last Update: 30 November 1993

function [ avwlen ] = huffcod(q,p)

%%%

% INPUT THE SYMBOLS TO BE CODED %

%%%

% INPUT THE NUMBER OF SYMBOLS TO BE CODED. NO TRIVIAL SOLUTION ALLOWED.

%q=0; % q = number of symbols. Set to 0 to ensure that the loop

% will be executed at least once

%while q<3 % Need at least 3 symbols for a non-trivial solution

%q=input('Enter the number of symbols: ');

if q<3,disp('Trivial solution. Use a larger number of symbols. '), end

%end

% ENTER THE SYMBOL PROBABILITIES.

% Note: The probabilities must sum to 1.00 and must be entered in

% descending order for the algorithm to work properly. Since the algorithm

% will give erroneous results if these errors are overlooked, error checking

% routines are included in later steps.

%disp(' ')

%disp('Enter the symbol probabilities ( in descending order).')

%for i=1:q, p(i)=input([' Enter the probability of s',int2str(i),': ']); end

% ENSURE THERE ARE ENOUGH PROBABILITIES ENTERED

% If <RETURN> is inadvertently struck before a probability is entered the

% input command could yield a probability vector which is too small. This

% causes the program to crash. This procedure prevents this from happening

% by setting all of the missing probabilities to zero. In this event the

% user can correct the wrong probabilities in a later step.

if length(p)<q, p=[p;zeros(q-length(p),1)]; end

% ERROR CHECK THE SYMBOL PROBABILITIES

correct='n'; % correct = 'n' ensures at least once through the error checking

% loop.

count=0; % count = 0 makes the loop a little simpler. It prevents the

% program from prompting for a correction until the loop has

% been executed at least once.

while correct ~= 'y' % Keep looping until correct.

if count>0; % This procedure will be executed only if there are errors

% to be corrected.

s=input('Enter the index of the incorrect probability: ');

p(s)=input(['Enter the correct probability for s',int2str(s),': ']);

end

count=1;

% Display the table.

disp(' ')

disp('Index Symbol Probability')

disp('-----')

for i=1:q

is=[int2str(i) blanks(6)]; is=is(1:7); % makes a string from the index.

ps=[num2str(p(i)) '000000']; ps=ps(1:6); % makes a string from the prob.

```

disp([' 'is,' s',is,' ',ps]) % displays the table
end
if abs(sum(p)-1)>1e-8 % Ensures probabilities sum to one.
    correct = 'n'; %tinha um "beep," antes
    disp(' ')
    disp('Error --> Probabilities do not sum to 1.00!')
elseif max(diff(p))>0 % Ensures probabilities are in descending order.
    correct = 'n'; %tinha um "beep" antes
    disp(' ')
    disp('Error --> Probabilities are not in descending order!')
else correct=input('Is the table correct? (Enter y or n): ','s');
    % Asks the user to verify that all the probabilities are correctly
    % entered. A 'n' response will prompt the user for corrections.
end
end, clear correct is ps count
p=p'; % p must be a column vector
pp=p; % pp = extra copy of the original probability vector
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% FORM THE Q-2 PERMUTATION MATRICES (LEFT MULTIPLICATION) %
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% INITIALIZE EACH MATRIX TO THE ZERO MATRIX OF APPROPRIATE DIMENSION
for i=1:q-2, eval(['P' int2str(i) '=zeros(q-i,q-i+1);']), end
% SUM THE LOWEST TWO PROBABILITIES AND DETERMINE NEW SORTED LOCATIONS
for k=1:q-2 % do for each of the q-2 permutation matrices
    Sum=p(q+1-k)+p(q-k); % sum the two lowest (and smallest) probabilities
    i=1;
    while Sum < p(i) % find highest location in p the vector for the sum
        eval(['P' int2str(k) '(i,i) = 1;'])
        i=i+1;
    end
    eval(['P' int2str(k) '(i,q-k:q-k+1) = [1 1];']) % This is the spot
    while i<q-k % form rest of matrix with the remaining probabilities
        i=i+1;
        eval(['P' int2str(k) '(i,i-1) = 1;'])
    end
    p=eval(['P' int2str(k)])*p; % multiply permutation matrix and probability
    % vector to get new probability vector.
end, clear p Sum k
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% FORM THE SYMBOLS %
%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%
% The symbols are formed using matrices of characters. The characters are
% ones, zeros, and blanks. Each row in a matrix represents a codeword. The
% final codewords are in the s0 matrix. Blanks are included in the matrices
% in order to make this part of the algorithm work efficiently. These blanks
% are removed in a later step.
% INITIALIZE ALL CODEWORD MATRICES TO BLANKS (Blank = 32 in ASCII)
for i=1:q-1, eval(['s' int2str(i-1) '= 32*ones(q-i+1,q-i);']), end
% SET RIGHTMOST CODEWORD VECTOR TO ['O' '1'] (0=48 in ASCII, 1=49 in ASCII)
eval(['s' int2str(q-2) '= [48; 49];'])
% WORK FROM RIGHT TO LEFT USING THE P MATRICES TO FORM THE CODEWORDS
% The codewords are formed from matrices of zeros (ASCII 48), ones (ASCII 49),

```

```

% or blanks (ASCII 32). Sq-1 is the rightmost matrix and has the [0 1]
% matrix. s0 is the leftmost matrix and contains the final codewords
% (except for extra blanks).
for i=q-2:-1:1
    twosum=find((sum((eval(['P' int2str(i)]))'))==2);
    % twosum is the index of the row of the permutation matrix with two ones.
    % This is the row which accomplishes the addition of the two lowest
    % probabilities. Its index indicates where the sum is to be placed in the
    % new probability vector. This index also gives information on how to
    % form the codewords.
    onesum=find((sum((eval(['P' int2str(i)]))'))==1);
    % onesum has the indices of all the rows of the permutation matrix with
    % only single ones. The indices indicate how the probabilities will be
    % placed in the new probability vector. These indices also give
    % information on how to form the codewords.
    eval(['s' int2str(i-1) '(1:q-i-1,1:q-i-1)=s' int2str(i) '(onesum,1:q-i-1);'])
    eval(['s' int2str(i-1) '(q-i ,1:q-i-1)=s' int2str(i) '(twosum,1:q-i-1);'])
    eval(['s' int2str(i-1) '(q-i+1,1:q-i-1)=s' int2str(i) '(twosum,1:q-i-1);'])
    eval(['s' int2str(i-1) '(q-i ,q-i)=48;'])
    eval(['s' int2str(i-1) '(q-i+1,q-i)=49;'])
    % The five lines above place the appropriate ones, zeros, and blanks in the
    % codeword matrices as the progression moves from the right to the left.
    eval(['clear P' int2str(i) ' s' int2str(i)])
end, clear onesum twosum
% FIND AND REMOVE THE BLANKS FROM EACH CODEWORD AND COMPUTE WORD
LENGTHS
for i=1:q
    eval(['S' int2str(i) '=(s0(i,:));']) % s0 has all the needed information
    eval(['c=find(S' int2str(i) '== 32);']) % find all the blanks
    eval(['S' int2str(i) '(c) = [];']) % remove all the blanks
    eval(['S' int2str(i) ' = setstr(S' int2str(i) ');']) % convert from ASCII
    eval(['L(i)=length(S' int2str(i) ');']) % compute the length of each codeword
end, clear s0 c
avwlen = sum(L*pp);
%%%%%%%%%%%%%%
% DISPLAY THE OUTPUT %
%%%%%%%%%%%%%%
disp(' ')
disp('Symbol   Probability   Code Word')
disp('-----')
for i=1:q
    is=[int2str(i) blanks(6)]; is=is(1:7);
    ps=[num2str(pp(i)) '000000']; ps=ps(1:6);
    disp([' s',is,' ',ps,' ',eval(['S' int2str(i)])])
end, clear is ps i q, disp(' ')
% COMPUTE AND DISPLAY AVERAGE WORD LENGTH
L_avg=sum(L*pp);
disp(['The average word length is ', num2str(L_avg)])
% COMPUTE AND DISPLAY THE ENTROPY
H=sum(pp.*log2(1./pp));
disp(['The entropy is ', num2str(H)])
% COMPUTE AND DISPLAY VARIANCE
var=sum(((L_avg-L).^2)*pp);
disp(['The variance is ', num2str(var)])

```



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